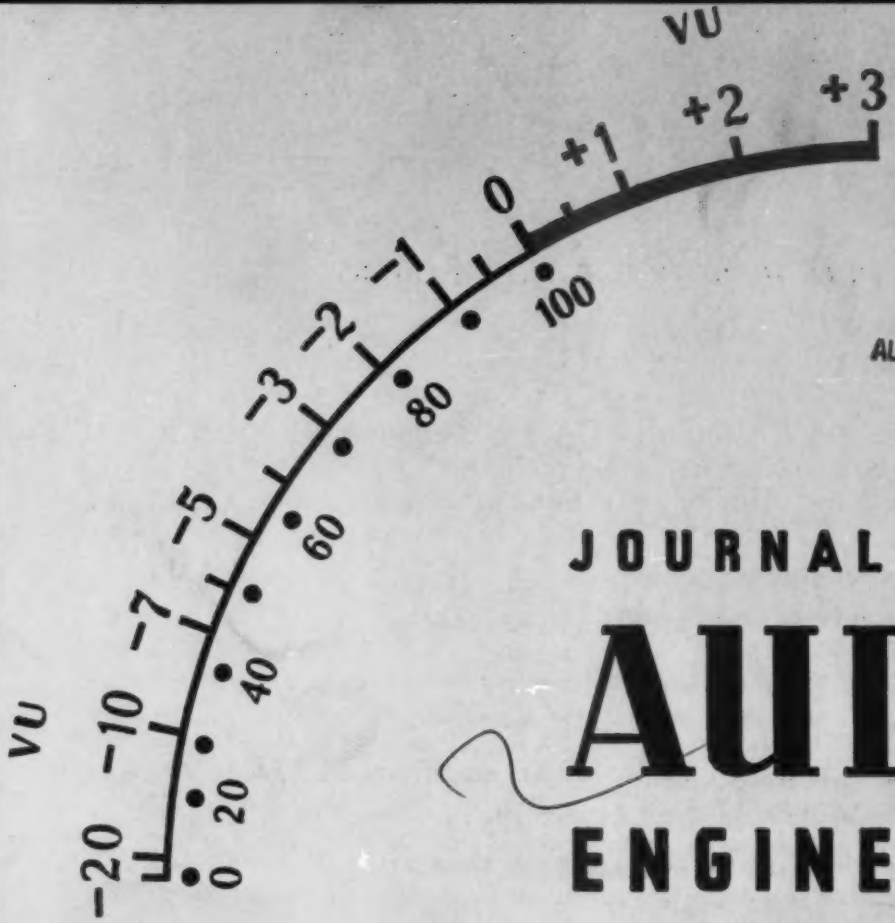

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VOLUME 7 NUMBER 3 JULY 1959

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Corner Speaker Placement*

PAUL W. KLIPSCH

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That superior sound quality is afforded by corner speaker placement has been recognized for more than thirty years. Measurements and listening tests were conducted with both monophonic and stereophonic reproducing equipment to compare the performance of speakers designed for use away from a corner and for corner installation. Measurable and audible advantages accrue with corner placement of either speaker, with the greater improvement being shown by a speaker of proper design. The advantages are more easily observed for stereophonic than for monophonic reproduction.

PART ONE: MONOPHONIC SOUND REPRODUCTION

Introduction

THE ORIGIN of the idea that there is an advantage to corner speaker placement may be as old as loudspeakers themselves. At any rate, patents on corner speakers date back to 1925.[1]

In the following investigation, a non-corner speaker with 6.7 cubic feet enclosure was tested out of and in a corner. With corner placement the bass response was improved more than 20 db. Listening tests confirmed the evidence of the measurement and indicated a further improvement when a speaker designed for corner installation was observed.

The reasons for improved sound quality are obvious: the smaller treble radiation angle required to cover a room area is one; the mirror image effect of the walls, increasing the radiation resistance, is another; a third is the fact that most well designed speakers offer a 90-degree radiation angle, and only from a corner can this radiation cover the entire room.

No information has been discovered giving the quantitative advantages of corner placement.

Experiment

An experiment was set up using a 6.7 cubic foot box with a 3-way drive system including a heavy-magnet 15-inch woofer motor. This unit will be referred to simply as "the box."

This box was placed on an open legged bench about 14 inches high and about 4 feet from each wall at a corner. The drive unit was about 4 feet from the floor. This was to deny benefit of wall and floor reflections. Two microphones were set up, one in the center of the room, and the other in the middle of the half of the room where the speaker was located. See Fig. 1.

A response pressure curve was run and plotted as Curve 1 in Fig. 2. Peaks and dips were sought, and no attempt to smooth by moving the microphones was attempted. The two microphones were averaged by rectifying and then adding. Indicated sound was held constant by varying speaker input, the input level being read off a calibrated potential divider between the oscillator and amplifier.

Oscillographic examination of the output of each microphone before rectification was used to detect harmonic distortion.

The microphones were calibrated by pistonphone and were down 0.5 db at 30 c relative to 300.

A considerable peak-to-trough ratio is normally to be expected from this sort of measurement. Some troughs in the response curve could have been raised by moving one or both microphones. The standing wave pattern was ignored, however, for the concern is with the relative performance of a given speaker in a given room, the independent variable being the location in the room. Outdoor response curves are more valuable for determining intrinsic speaker performance; here the problem is to evaluate the advantages of different locations of a given speaker.

The speaker was then set on the floor over the same spot in the room. The microphones were not moved. The pressure response is Curve 2 of Fig. 2.

Finally the speaker was moved into the corner and the response is Curve 3 of Fig. 2.

For the purpose of these tests, various microphone placements were contemplated. Experience indicated that the extreme bass range could be favored by microphone placement in a far corner or in close proximity to the speaker. Several microphone locations were tried with the oscillator swept quickly through the pertinent spectrum, and the two-

* Received April 24, 1958. Revised copy received July 18 and September 17, 1958.

[1] M. Weil, U. S. Patent 1,820,996 issued 1931, filed 1925; and Sandeman, U. S. Patent 1,984,550 issued 1934, filed 1931.

microphone placement chosen was the one giving the lowest peak-to-trough ratio for all three speaker locations.

Movement of the speaker from one test to another was only about 5 feet, or ¼ wavelength at 50 c, not enough to affect the measurements materially—say more than 3 or 4 db. The closer speaker-to-microphone spacing of Locations 1 and 2 would favor the bass for these locations, compared to Location 3. The listening tests qualitatively corroborated the measurements, the differences being gross enough to discern without the refinement of an "A-B test."

Taking the speaker down off its perch improved the response at 50 c more than 20 db. On legs, the speaker cutoff should be considered to be 65 c. On the floor the response at 50 c is 10 db down from its peak at 65 c.

In the corner, however, the useful range extends to 38 c. The improvement in response at 40 c is 9 db, which represents an 8-fold increase in efficiency.

In each step toward better placement, the peak-to-trough ratio in the usable spectrum was reduced, and the distortion was reduced by a large amount.

While the major aspect may be thought to be the improved and extended bass, the reduced distortion is far more important. Less noticeable on the curves is the reduced peak-to-trough ratio (the lowered effect of standing waves in the room) but this effect is actually of considerable importance. More about this will be given in the section **Listening Tests.**

All these benefits are achieved for a non-corner-type speaker merely by placing the speaker in a corner, in spite of the fact that the speaker was deliberately chosen as a non-corner type.

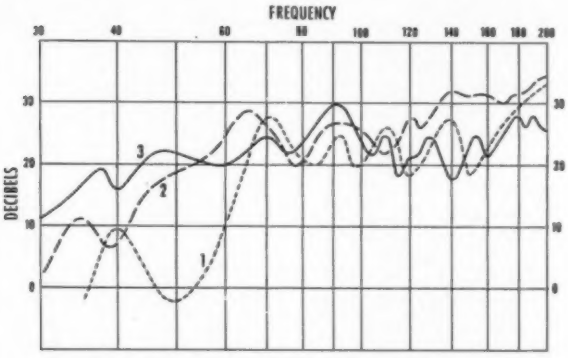

FIG. 1. Plan view of speaker and microphone location in tests. Speaker location 1, as shown, was on a 14-in. high bench to deny benefit of the mirror image formed by the floor. Speaker location 2 was at the same plan location but on the floor to couple the actual speaker with its mirror image. Location 3 was on the floor in the corner to couple the actual speaker to all its mirror images.

Microphones were 6½ feet apart, located as shown; their outputs were rectified and then added to produce an averaging effect and to reduce standing wave effects.

Distortion, indicated by the cathode ray scope, was large below 70 c for the conditions of Curve 1, considerably below 60 c for conditions of Curve 2, and slightly below 35 c for conditions of Curve 3.

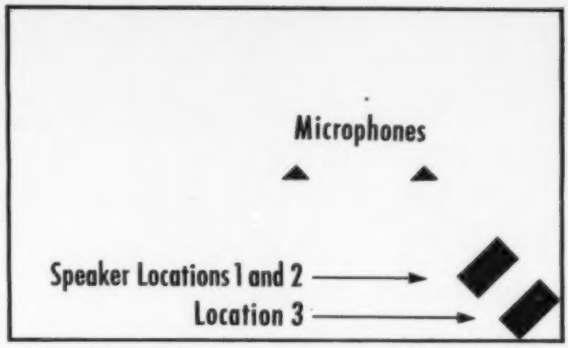

FIG. 2. Response Curves 1, 2 and 3 correspond to the speaker locations 1, 2 and 3 in Fig. 1.

The advantages of corner placement are:

reduced distortion due to better loading;

increased bass range for a given size of speaker system;

increased bass range for a given size of room, or increased effective size of the room;[2]

smoother response—less peaking in the second octave.

In addition to benefits in the bass range, other benefits derive from corner placement:

the available radiation angle of high-frequency system covers the whole room area (most speakers of good quality offer a 90-degree radiation angle);

better ratio of direct to reverberant sound; longer travel of a wave before it strikes a reflecting surface;[3] better appearance due to smaller size for a stated performance.

The above observations are applicable both to monophonic and stereophonic reproduction.

All these advantages appear as a result of corner placement of any speaker, but of course a maximization of all the advantages is obtained by using a speaker type designed for corner operation.

Listening Tests

The curves and objective data ignore frequencies above 200 c, which are not particularly affected by corner placement except as it affects room coverage with a given radiation angle.

For a listening test, however, the middle and upper ranges are all important, so a three-way drive system was used.

The box was subjected to a listening test in the various locations. An A-B test was not necessary to reveal the superiority of the floor location compared to the location on legs. Moving from floor to corner afforded as great a listen-

[2] Paul W. Klipsch, "Room Dimensions for Optimum Listening and the Half-Room Principle," *Trans. IRE, PGA,* **AU-6**, 1 (January-February, 1958).

[3] *Ibid.*

ing difference as did moving from bench to floor, especially where heavy bass comprised a considerable part of the program material.

Finally a comparison was made between the box and a corner-horn back-loading system of the same front panel size, the same drive system, and some 22 per cent less occupied space.[4]

The advantage of the unit designed for corner application was immediately evident. The improvement in bass over that reproduced by the box was almost as great as the improvement afforded by moving the box from the floor to the corner.

Conclusion

These experiments should illustrate the fact that good acoustical design takes maximum advantage of the mirror images produced by the floor and walls, and that the maximum benefits are derived when the mirror surfaces are proximate to the prime source. This excludes styling which involves legs or risers which separate a speaker's radiating area by even as little as half a foot from the floor; it dictates structure which places the radiation surfaces as close to the walls and floor as possible.

PART TWO: STEREOPHONIC SOUND REPRODUCTION

Introduction

A SERIES of experiments was undertaken to determine preferred speaker separation for two-channel stereophonic sound reproduction.

First an outdoor experiment discloses that an angle as small as 9 degrees suffices to make the stereo effect discernible, and desirable angles are of the order of 30 to 40 degrees for most observers.

Indoors, the angles must increase to nearly 30 degrees to afford a detectable stereo effect, and to 50 degrees or more to be desirable.

A narrow margin of listener preference was found for wide speaker spacing compared to narrow spacing in two-channel arrays. A preponderance of preference was shown for corner placement. Three-channel stereo with wide spacing was preferred over all other arrays tried.

Experiment One

In an outdoor environment, what minimum stereo angle is required that the stereo effect be observed, and what is the preferred angle?

Two small speakers were placed 8 feet apart against a wall, about 2 feet above the ground, outdoors.[5] Program material was a high quality recording of a symphonic band. Observers were asked to explore the median line between speakers, and indicate: (1) the preferred distance from the speakers, and (2) the maximum distance at which they could detect the stereo effect.

Experience indoors indicated an *a priori* estimate that the minimum stereo angle would be about 25 or 30 degrees. The following results were observed:

		Preferred Angle (in degrees)	Minimum Angle (in degrees)
5 Men	Max.	70	16
	Aver.	45	13
	Min.	28	11
3 Women	Max.	33	14
	Aver.	26	12
	Min.	19	9
Average, Men and Women		37	12

The average minimum angle was 12 degrees instead of the expected 25 degrees or more. A more detailed experiment might well bring this angle considerably below 12 degrees. Note, however, this is *outdoors* and the same experiment conducted indoors produced data which put the minimum angle much larger. This is as would be expected; room reverberation masked the direct sound and a listener had to subtend a larger angle to detect the stereo effect. Another factor involved was that the ambient noise level was extremely low compared to an average city dwelling.

But the results remain: preferred listening subtended angle, 37 degrees; minimum 12 degrees, for outdoor stereo (average figures).

Experiment Two

Indoors, what are the preferred and minimum angles?

In a room 16 by 25 feet, one pair of speakers was set up 7 feet apart, and another identical pair at about 14 feet.

The array was disposed along the wall at one end of the room. The row of speakers was about 2 feet from the 16-foot wall and the end speakers were about 1 foot from the long walls. The spacing from the walls was to avoid any benefit from the improved bass range due to corner effects. It was desired to find the preference for the narrow or the wide spacing, not whether the improved bass range afforded by the corner would induce a preference.

The inside pair was fed from one stereo-pair of amplifiers, the outside pair from another stereo-pair of amplifiers, and the volumes were carefully adjusted to be the same. Switching was between the tape preamplifiers and the stereo-pair of power amplifiers.

[4] While the box had a volume of 6.7 cubic feet (about 7.0 cubic feet outside), its chargeable space included 3 cubic feet of unused space behind it. Thus the total space taken up in the corner was approximately 10 cubic feet. The corner horn used for comparison requires only 7.8 cubic feet of space.

[5] Small for portability, and to determine various configurations. Speakers were 3-way with the "K-ORTHO-12-A" drive system, with a peak-to-trough ratio less than 10 db from 150 to 21,000 c. Mid-range and tweeter were of the horn-loaded type to minimize distortion.

The listener was first seated at a distance of 15 feet on a median line between the speakers, and asked to indicate a preference.

The results:

Preference for narrow spacing	1
Indecisive, but generally favoring narrow spacing	2
Indecisive, but generally favoring wide spacing	3
Preferred wide spacing	1
Changed opinion from preference of narrow to preference of wide spacing	1

This is almost a standoff. Note the subtended angles here are 50 degrees and 27 degrees for the wide and narrow spacings.

The angular values found are consistent with those which a concert listener would observe in, say, the 5th to the 10th row; the orchestra might be spread out 45 feet (typically) and the observer would sit in a distance range of 20 feet (front row), 45 feet (more or less ideal) and 90 feet in the back of the balcony. A larger auditorium would be likely to have a wider stage. Expressed in degrees, the optimum listening stereo angle in the auditorium would range from over 90 degrees down to perhaps 45 degrees depending on the preference for front row or far balcony.

Next with the same speaker configuration, the observers were asked to "explore" the end of the room away from the speakers (about 22 feet) and indicate if they could detect the stereo effect.

Only one observer could detect the stereo effect at a distance of 24 feet with the 7-foot spacing, but all could with 14-foot spacing.

Experiment Three

What is the effect of using one speaker in the corner?

The widely spaced experiment was repeated with one change: one small speaker was removed and a large full-bass-range corner horn speaker substituted. The spacing increased to 15 feet, but the stereo geometry seemed to condense in that the angle seemed to become less rather than greater, yet the stereo effect was still discernible at the far end of the room. Also the "two-peephole effect" of the array using small speakers was brought to attention by its disappearance when using a large corner speaker. The placement in the corner seems to aid in completing a curtain of sound. This is believed to be due to the simpler mirror image system pertaining to the corner type speaker.

Most narrow-spaced stereo systems are also narrow band, in that the speakers are too small to exhibit an extended bass response. With an added bass range plus the added reverberation due to such added bass, it is a considered opinion that a wider speaker spacing would be highly desirable if not absolutely necessary. The larger effective radiating surface of the full-range corner speaker more nearly simulates the original sound spatially as well as tonally and therefore approaches a stereophonic effect. The smaller point source speakers tend to focus the sound into two discrete points; large corner speakers offer a closer approach to a curtain of sound, while still focusing individual entities in the mass of original sound.

Experiment Four

For non-corner speakers, is the long or short wall of an oblong room preferred?

A near-corner array on the short wall of the room (the 14-foot spacing of Experiment 2) was compared with a similar array on the long wall of the room, with about 24-foot spacing. Again the small speakers were employed. The preference for the 14-foot spacing on the short wall of the room ranged from mild to positive with only a couple of observers favoring the long wall placement.

This would appear to temper my earlier conclusions.[6] The fact that on some types of music some listeners perceive an impression of a "hole in the middle" is not to be overlooked. The "hole in the middle" effect is less likely to be noticed when the speakers are placed in the corners at the narrow end of the room.[7]

Experiment Five

This is not a formal experiment set up in a laboratory, but is a collection of this writer's observations made over the period since the fall of 1956 when the idea was evolved of three-channel stereo derived from two-track source material.[8]

A very informal comparison was made between two-channel stereo using two corner speakers on the 16-foot wall, and three-channel using two corner speakers on the 25-foot wall plus a third speaker in the middle of the 25-foot wall.

A great many demonstrations were made in which the middle speaker could be readily cut in and out of circuit. Here the preference was overwhelmingly for the three-channel with the long-wall spacing.

One situation with two-channel stereo on wide corner

---

[6] Paul W. Klipsch, "Experiences in Stereophony," *Audio*, **39**, 16 (1955); and

Paul W. Plipsch, "Making Stereophonic Tapes," *Hi-Fi Music at Home*, **2**, 54 (1955).

[7] A configuration suggested in the sheet accompanying RCA-Victor stereo tapes.

[8] Paul W. Klipsch, "Stereophonic Sound with Two Tracks, Three Channels by Means of a Phantom Circuit," *J. Audio Eng. Soc.*, **6**, 118 (1958).

Also popular digests by the same author: "Two-Track, Three-Channel Stereo," *Audiocraft*, **2**, 26 (1957); and "Phantom Channel," *Audiocraft*, **3**, 37 (1958).

Note that this does not require three sound tracks, but recovers a third channel from two sound tracks. This depends on the fact that if two microphones are properly spaced relative to each other and to the plane of the sound source, their combined output is that of a third microphone in the middle. This middle microphone—the microphone that wasn't there—can be recovered and reproduced by combination of the two accessible sound tracks. All stereo recordings so far played over this writer's two-track three-channel system have displayed the focusing effect of the derived center channel.

Continued on page 114

Recent Cone Speaker Developments*

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This paper discusses the factors governing the low-frequency response and distortion of cone speakers. Laboratory techniques for the evaluation of cone compliances are described. A cone speaker design using a new linear compliance is discussed in detail, and acoustic measurements on this speaker are presented.

MATHEMATICAL CONSIDERATIONS

TODAY'S high quality program material available from records, tapes, and FM radio stations has sped up the development of wide-range commercial speaker systems which become comparable in quality with early non-commercial designs, such as the well known Fletcher System developed at Bell Telephone Laboratories around 1934. This paper discusses the problems of the reproduction of low frequencies with cone speakers.

The performance of a cone speaker at low frequencies can be described accurately by comparatively simple mathematical formulas. Fig. 1 shows the equivalent electrical circuit of a cone speaker mounted in an infinite baffle driven from an amplifier with open circuit voltage e and source impedance R_G. All elements have been transformed into the mechanical side of the speaker. In this diagram $e\frac{Bl}{R_E}$ is the transformed open circuit voltage, $\frac{B^2l^2}{R_E}$ the transformed resistive components of the electrical side, consisting of the source impedance R_G and the voice coil resistance R_{VC}. For this low-frequency investigation the inductive component of the voice coil impedance can be neglected. The rest of the circuit consists of the mechanical and acoustical elements of the cone speaker. R_M represents the mechanical losses; L_C, the total moving mass of the speaker; C, the total compliance of the suspension system; L_A, the radiation mass; R_A, the radiation resistance. In the frequency range to be discussed L_A can be assumed to be constant and R_A will increase with the square of the frequency. B is the flux density and l the length of the voice coil conductor in the magnetic field.

At resonance frequency this circuit can be simplified as shown in Fig. 2a. At resonance the reactive components cancel each other. It is assumed that the mechanical losses are small compared with the damping provided by the voice coil source circuit. To obtain simple expression for the power radiated, which in the equivalent circuit would be the power delivered into the radiation resistance $K\omega_0^2$, it is further assumed that the radiation resistance is small compared with $\frac{B^2l^2}{R_E}$. Eq. (1), in Fig. 2a, shows the expression for the radiated power. At the first glance this formula looks rather surprising, as it indicates that the radiated power increases as Bl decreases. This relation exists, however, only as long as the assumptions made to derive this simple formula are valid. These assumptions

$$\left(\frac{B^2l^2}{R_E} >> R_M, \frac{B^2l^2}{R_E} >> K\omega_0^2\right)$$

were that at resonance the damping is mainly electrical, because of high Bl, low voice coil losses, and small source resistance of the amplifier. Most of today's woofers, except a few types with extremely low efficiency, will meet this condition.

At mid-frequencies the circuit can be simplified as shown in Fig. 2b. The controlling element is the total mass, L, of the system consisting of the speaker mass and the radiation mass. It should be remembered in this connection that a flat response is obtained from a piston in an infinite baffle over the frequency range where the piston dimensions are small compared with wave length, provided the movement of the piston is mass controlled. Mass control means that velocity decreases linearly with increasing frequency, thereby compensating for the radiation resistance, which increases with the square of the frequency. For the sake of simple end results, it is again assumed that the radiation resistance is small compared with the reactive mass component. Eq. (2) shows the acoustic power radiated under these assumptions. It should be noted that this expression does not contain the frequency, indicating flat response in this range. The prescriptions for high efficiency at mid-frequencies, as indicated by Eq. (2) are high Bl (large magnet), small R_E (low voice coil losses), and small moving mass. The first

* Presented February 19, 1959 at the Sixth Annual Western Convention of the Audio Engineering Society, Los Angeles.

† Chief Engineer.

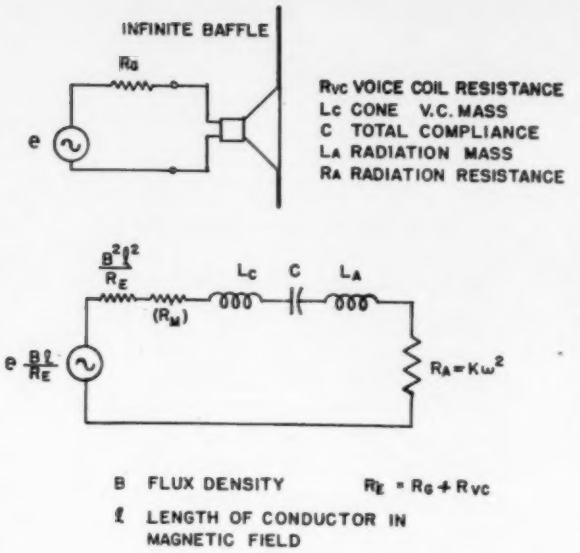


FIG. 1.

requirement is just opposite the requirements for high efficiency at resonance, and low voice coil losses maintaining high l contradicts the requirement for small moving mass. The design problem will be to find the optimum compromise.

Eq. 3 in Fig. 2 shows the ratio of radiated power at resonance frequency to radiated power at mid-frequencies. This ratio depends, in first approximation, on the Q of the system. To obtain a flat response down to the resonance frequency a Q of 1 would be required. This means, however, that the system will be less than critically damped and will therefore have oscillatory transient distortion. For a critically damped system $Q = .5$, the response at resonance frequency will be 6 db down from the mid-frequency range.

The above over-simplifications were used in order to obtain end results which indicate the controlling factors without obscuring details. It can be shown by rigorous analysis that the relations indicated by above equations are correct (e.g., Leo L. Beranek, *Acoustics*, McGraw-Hill Book Company, Inc.). As a further proof, Fig. 3 shows a measurement made on a 15-in. woofer mounted in a 10 cubic foot closed box. In this instance the resonance frequency is doubled by the enclosure. The amplifier source impedance was adjusted so that the Q of the system was .5 as indicated by the impedance curve (lower curve in Fig. 3). The difference in frequency response between resonance frequency and 150 c is indeed very close to 6 db. The further rise of sound pressure above 150 c is due to the fact that the radiation begins to become directive in this range.

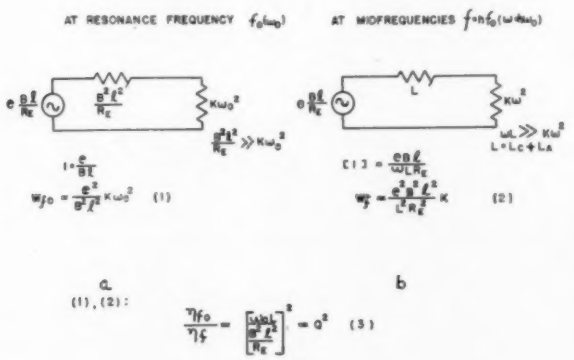


FIG. 2.

DESIGN REQUIREMENTS

The theoretical analysis indicates the following requirements for the design of a good low-frequency speaker:

A. The total moving mass has to be as small as possible for high mid-frequency power. Limitations to this requirement are the stability of the cone and the requirements for a low resonant frequency, which, with a very small mass, would result in a too soft and unstable suspension system. A practical minimum mass for a 15-in. woofer is approximately 20 g, which permits a resonance frequency of 22 c with sufficiently stable suspension system.

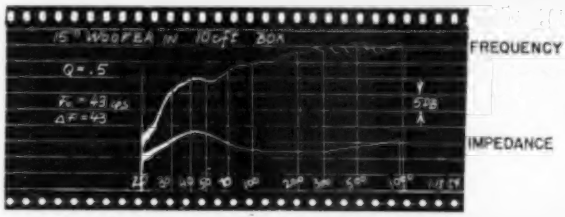


FIG. 3.

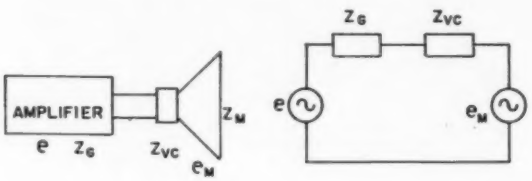
B. Another requirement for high mid-frequency power is that Bl should be large and the losses in the voice coil should be small. This calls for heavy magnet and voice coil. The choice of the magnet size will be influenced by economic factors, and the size of the voice coil has to be compromised with the requirement for a small total mass and adequate high-frequency response of the speaker.

C. Eq. (3) Fig. 2 indicates that the low-frequency response will depend on the Q of the speaker system. If critical damping is chosen (smallest damping for non-oscillatory transients) the frequency response will be 6 db down at resonance frequency.

It should be pointed out that the Q defined in Eq. (3) is the Q of the whole system, including the source resistance. The source resistance required for a certain Q will vary between different speakers, depending on their efficiency and mechanical-acoustical damping.

It should also be noted that with proper control of Q by R_E good low-frequency response can be obtained without the necessity of high mass and low Bl and consequently low mid-range efficiency.

A good cone speaker must not only have flat response down to low frequency, but also be able to produce low frequency with low distortion. This means that at low frequencies the cone has to move with large amplitudes proportional to the driving current in the voice coil. This can be expressed simply by saying that the whole system must be linear. In detail this requires that the voice coil stay in a uniform magnetic field for the whole length of travel and that the suspension system be linear and symmetric for the maximum amplitude. The first requirement must be met since the driving force is generated by the interaction of the voice coil current with the magnet field, and this force must be proportional to the voice coil current. The second requirement can be relaxed if special amplifiers with negative impedance are used.



$$e = I(Z_G + Z_{VC}) + e_M \qquad (1)$$

$$e_M = Bl v\,, \quad v = \frac{F}{Z_M}\,, \quad F = Bl I \qquad (2)$$

$$(2) \quad e_M = \frac{B^2 l^2 I}{Z_M} \qquad (3)$$

$$(1)(3) \quad e = I\left(Z_G + Z_{VC} + \frac{B^2 l^2}{Z_M}\right) \qquad (4)$$

$$Z_G = -Z_{VC} \qquad (5)$$

$$(4)(5) \quad I = \frac{e\, Z_M}{B^2 l^2} \qquad (6)$$

$$(6) \quad v = \frac{e}{Bl} \qquad (7)$$

Fig. 4.

The principles underlying the use of negative source impedance are indicated in Fig. 4. A speaker with the voice coil impedance Z_{vc} and the mechanical impedance Z_M is driven by an amplifier with the open circuit voltage e and source impedance Z_G. Eq. (1) is the loop equation for the system. The open circuit voltage e has to be equal to the voltage drop across the source impedance and voice coil impedance plus the voltage e_m developed by the movement of the voice coil. Eq. (2) and (3) indicate how e_m can be expressed in terms of the driving current I and the mechanical impedance of the speaker. Substituting Eq. (3) into (1) gives an equation between the open circuit voltage and the current of the system. The last term in the bracket is the magnitude of mechanical impedance as reflected into the electrical circuit. Assuming now that the source impedance is equal to the negative voice coil impedance (Eq. 5), Eq. (4) reduces to Eq. (6). Combining (6) with Eq. (2) we find that the velocity of the system becomes independent of the mechanical impedance Z_m, which means that as long as Bl is constant throughout the range of excursion the movement of the system will be linear regardless of any non-linearity of the mechanical system. This measure, however, has some undesirable influences on the frequency

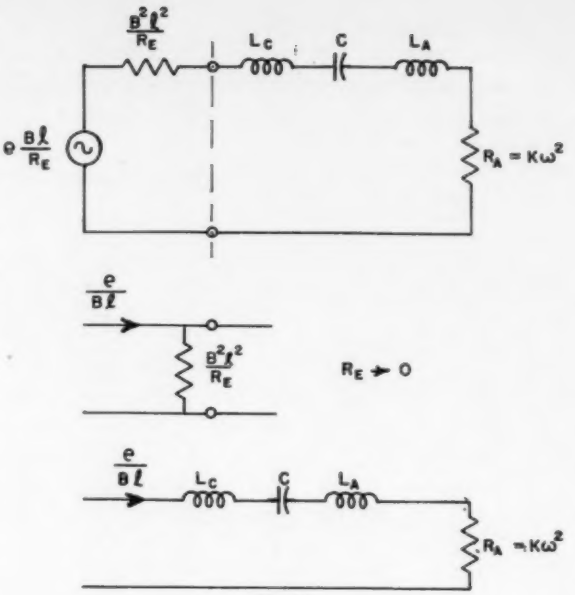
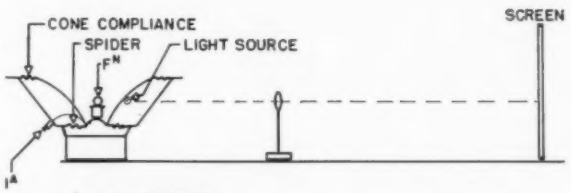


Fig. 5.

MEASUREMENT OF: FORCE - DISPLACEMENT FUNCTION
CURRENT - DISPLACEMENT FUNCTION
AND Bl



Bl MEASUREMENT:
IF WEIGHT AND CURRENT ARE BALANCED:

$$F = Bl I\,, \qquad Bl = \frac{F^N}{I^A} \quad \text{MKS}$$

Fig. 6.

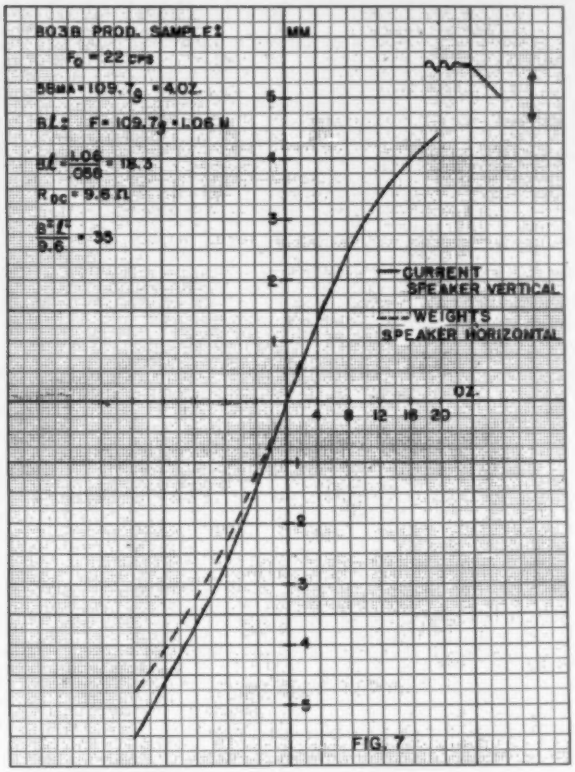


FIG. 7.

response of the speaker. Fig. 5 shows once more the equivalent circuit for the speaker-amplifier system. The source impedance is now equal and negative to the voice coil impedance, which means that R_E becomes 0. To see what that means we can transform the voltage source into a current source, as indicated in Fig. 5. It becomes quite apparent now that R_E going to 0 means that the circuit is driven by a constant current, which in mechanical terms means that the cone is moving with constant velocity. Since the radiation resistance will decrease with the square of the frequency as the frequency is decreasing, the frequency response of such an amplifier-speaker system will drop 6 db per octave with decreasing frequency. An amplifier which delivers one watt into the speaker at 320 c would have to deliver at 30 c a voltage equivalent to 100 w power delivered into the mid-range impedance if it is compensated for this dropping frequency response.

For most practical purposes it therefore appears to be more advantageous to develop a speaker with a linear suspension system than to compensate with negative amplifier impedance. This has become even more practical since new cone compliances made of phenolic impregnated textiles have recently become available. Fig. 6 shows how the complete speaker system or any of the suspension components can be evaluated statistically. The displacement is measured by optical means. A light source consisting of a miniature bulb is projected on a screen, allowing very accurate displacement measurement. The deflecting force can either be weights placed on the cone or, if the whole system

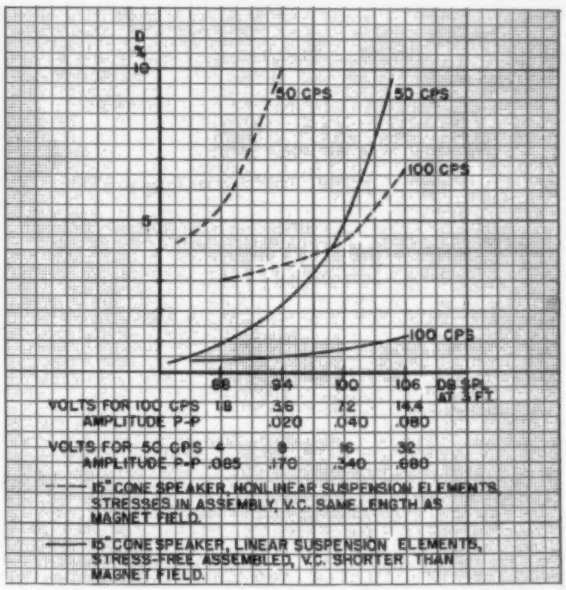

D
10
5
50 CPS
50 CPS
100 CPS
100 CPS
88 94 100 106 DB SPL AT 3 FT
VOLTS FOR 100 CPS AMPLITUDE P-P 1.8 3.6 7.2 14.4 .020 .040 .080
VOLTS FOR 50 CPS AMPLITUDE P-P .085 4 8 16 32 .170 .340 .680
15" CONE SPEAKER, NONLINEAR SUSPENSION ELEMENTS, STRESSES IN ASSEMBLY, V.C. SAME LENGTH AS MAGNET FIELD.
15" CONE SPEAKER, LINEAR SUSPENSION ELEMENTS, STRESS-FREE ASSEMBLED, V.C. SHORTER THAN MAGNET FIELD.

FIG. 9.

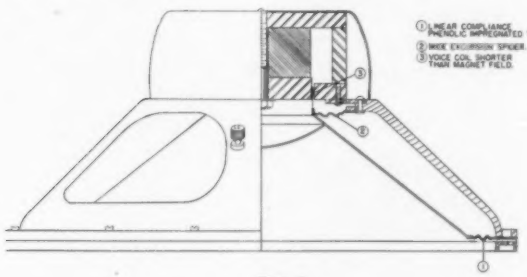


FIG. 8.

is to be tested, dc current through the voice coil. With this set-up it is also possible to measure very accurately the Bl product of the speaker. The cone is deflected by means of a weight placed upon it, and a current is then fed into the voice coil of such a magnitude that the cone is restored to its original position. Since the force from the voice coil equals the force from the weight, Bl can be computed as indicated in Fig. 6. Fig. 7 shows a graphical representation of measurements made with the described arrangement. The displacement of the cone is plotted *vs* the deflecting force, which can either be a weight or the current through the

voice coil. As can be seen, the current-displacement function is linear over a substantial range. This has been achieved by using suspension elements (1 and 2 in Fig. 8) which have a linear force displacement function over a wide displacement range, by stress-free assembly of those elements, and by accurately positioning the voice coil in a long magnetic field (3 in Fig. 8). Fig. 9, solid curves, shows the distortion of such a speaker in a 10 cubic foot closed box at 50 c and 100 c as a function of sound pressure level at 3 feet distance, as measured in an anechoic room. The graph also indicates the voltages and peak-to-peak amplitudes for the various S.P.L. The dashed curves in Fig. 9 show how rapidly distortions increase when above linearity requirements are not observed. The distortions shown by the dashed curves were measured on a speaker with a conventional paper compliance of limited linear excursion range. No jigs were used to assure stress-free assembly and correct voice coil position, and the voice coil had the same length as the magnet field.

CONCLUSION

If the requirements for good frequency response and linearity are observed it is possible with present day technology to build a good 12- or 15-in. woofer with resonance frequency around 22 c, good low frequency response, and low harmonic and transient distortion, without sacrifice in speaker efficiency. A speaker system built with these low-frequency components and large enough to take full advantage of the low resonance frequency will readily show up low-frequency turntable rumble and regulation thumps of inadequate compressors in some broadcasting programs. These new improved speakers now in turn require higher perfection of program sources.

Continued from page 109

spacing which is objectionable to some listeners is the reproduction of a soloist or an intimate group like a string quartet. The three-channel stereo restores such a group to its intimate dimensions, even with arrays subtending more than a 60-degree angle.

CONCLUSION

The following advantages of corner speaker placement pertain particularly to stereo:

Better stereo area coverage due to radiation angle being fully utilized,

Better stereo separation and larger stereo area,

Better focus of all events between flanks; better reproduction of stereo geometry,

Full vantage of the derived third channel which (properly used) offers precise focus and sharp definition of all sonic events between extreme flanks.

All these advantages appear as a result of corner placement of any speaker, but of course a maximization of all the advantages is possible by using a speaker type designed for corner operation.

Genuine stereophonic sound reproduction may be regarded as a close approach to recapturing the original spatial effect or "geometry," just as "high fidelity" aims to recapture the original tonal effects or "tonality." These cannot be achieved by merely pin-pointing the two halves of a musical group.

The more nearly genuine stereo is a more difficult and expensive thing to achieve. The end result should be a curtain of sound, not just a couple of peepholes.

The more sophisticated system will require larger and more expensive speakers, and will render a larger part of the room available for listening; a given sound will not be pin-pointed at the tweeter opening of a particular speaker but ideally will appear at some point or space in a curtain of sound between the speakers.

When one graduates to the more sophisticated stereo system, several opportunities open up; corner speaker placement becomes possible with all the advantages incident thereto; the third stereo channel becomes worthwhile. The results in listening are:

(1) a more nearly continuous curtain of sound;
(2) a larger effective area of sound curtain;
(3) a more accurate geometric delineation—as an extreme case, a string quartet will occupy a small central part of the sound curtain even with flanking corner speakers 25 to 50 feet apart;
(4) a simulation of the original sound and environment, including the ability of the members of the audience to change location over a wide area while the sound remains natural both in tonality and geometry.

Some Thoughts on Geometric Conditions in the Cutting and Playing of Stereodiscs and Their Influence on the Final Sound Picture*

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The influence of various geometric conditions in the cutting and reproducing process is investigated. It is found that stylus contour plays no important role. Axis orientation does play a role; any deviation herein causes cross modulation, ultimately leading to distortion of the original stereophonic soundpicture. Slant and rotation of planes of axes may be disregarded since the resultant difference angles are small enough to be neglected. Finally, the nature of soundpicture distortion caused by such cross modulation is investigated. Amplitude distortion leads to angular shift of the sound sources (panorama distortion), phase relationship of cross modulation components causing increase or decrease in width of the soundpicture (basis-distortion).

INTRODUCTION AND DEFINITIONS

THE OBSERVATIONS in this report are based on the use of the orthogonal method of two-signal modulation in one groove. The axes of modulation should be inclined 45° to the disk surface. It is furthermore always assumed that no inherent crosstalk exists in the cutter-head or playback cartridge proper.

It is defined that:

a) *Out-of-phase movements* L and R will result in *lateral* groove modulation.

b) *Lefthand* signal information will appear as the *inner sidewall* modulation (= movement L).

Pictorially, these definitions will be presented as sketched in Fig. 1.

In this report, phase and amplitude relationships are considered for mechanical movements, rather than electrical signals. It should be pointed out that out-of-phase movements (lateral or sideways modulation) will in practice be caused by electrical signals applied to the cutter head, which are by definition in phase.

Now deviations in symmetry or orthogonal orientation of axes will result in crosstalk between channels, ultimately leading to angular and width distortion of the reproduced stereo soundpicture as compared to the original one. These distortions are defined as *panorama* and *basis distortion* and will be dealt with in Section 4 of this report.

* Received January 11, 1959. Presented at the Tenth Annual Convention of the Audio Engineering Society, New York, September 30, 1958.

The deviations in geometric layout which are likely to be encountered in practice may be grouped as follows:

1. *Cutting-stylus contour*
   - 1.1 Cutting angle = 90°.
   - 1.2 Asymmetric.

   Symmetry is said to exist when the bisector of the cutting angle coincides with the normal to the disk surface.
2. *Axes of modulation or sensing.*
   - 2.1 Non-orthogonal, asymmetric.
   - 2.2 Orthogonal, asymmetric.
   - 2.3 Slanted and rotated plane of axes.

We will first investigate to what extent crosstalk may be expected as caused by the indicated geometric conditions. Subsequently, the influence of crosstalk on the reproduced sound picture will be considered.

THE INFLUENCE OF CUTTING STYLUS CONTOUR

1.1. *Cutting angle = 90°.*

The axes along which the cutterhead system exerts the modulation forces are supposed to be both orthogonal and symmetric. It is furthermore assumed that only one channel, for instance the righthand channel, is modulated, the other channel being idle.

From Fig. 2 it is evident that an obtuse or acute cutting angle will also cause some modulation to appear on the left-hand groove wall. This is in fact recorded crosstalk.

If such a groove is, however, traced by a stereo cartridge with perfectly orthogonal and symmetrical sensing axes, this crosstalk will not appear in the left hand reproducing chan-

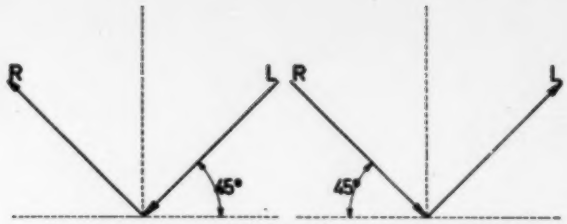


FIG. 1. Vector presentation of 45/45 recording method.

nel since the center of the playback stylus (pictured as a sphere in Fig. 2b) moves along a line inclined 45° to the disc surface. In general, whenever the orientation of the sensing axes coincides with that of the modulation-axes, no crosstalk results from the use of cutting styli having included angles other than 90°.

1.2. *Asymmetric contour*

In a rather similar way, the use of cutting styli with asymmetric contour will lead to cross modulation on the unmodulated groove wall (Fig. 3). Again, tracing by a cartridge with coincident orientation of sensing axes will prevent this cross modulation from appearing in the idle channel.

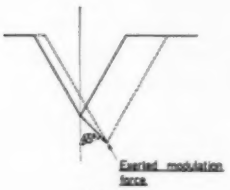


FIG. 2a

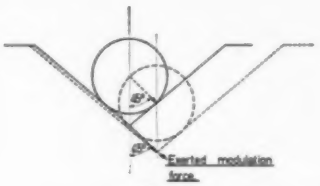


FIG. 2b

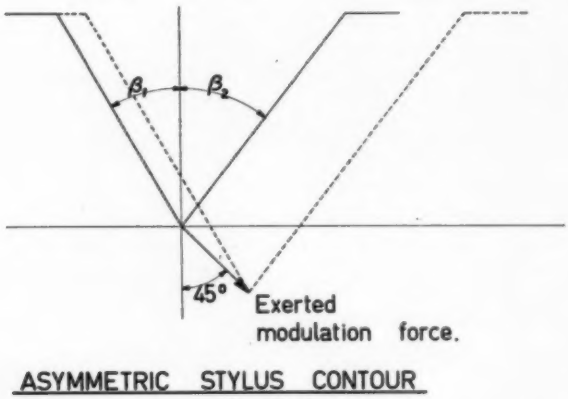


FIG. 3

The use of cutting styli, having included angles other than 90°, leads to modulation of the groove wall facing the idle recording system.

THE INFLUENCE OF AXIS-ORIENTATION

2.1. *Non-orthogonal, asymmetric orientation*

In order to arrive at a universal equation for the relation between the magnitudes of cross talk and signal component, the trigonometric calculation has been based upon the presumption that modulation axes do not coincide with sensing axes, either system being non-orthogonal and asymmetric.

In Fig. 4, let vectors AB and AC be representative for the movements of the cutter head systems, AD and AE being the directions of sensing axes.

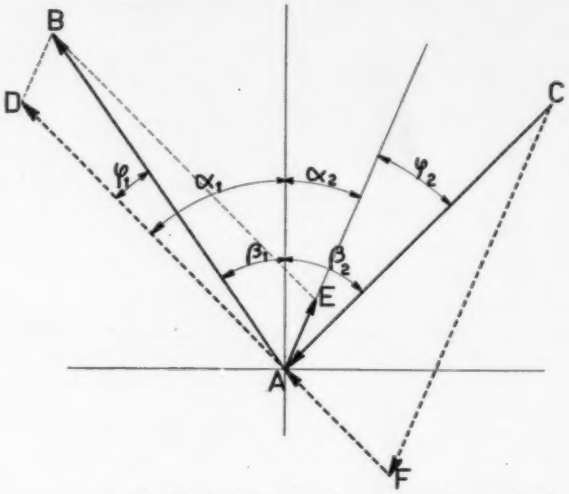


FIG. 4. Non-coincident orientation of axes results in crosstalk.

The angles between either AB, AC, AD or AE and the normal to the disc surface are indicated by β_1, β_2, a_1 and a_2, respectively.

It is evident that a single modulation vector AB will be sensed in both direction AD and AE. The relation in decibels between decomposed components AD and AE is given by:

$$\frac{AD}{AE} = 20 \log_{10} \frac{\sin (a_2 + \beta_1)}{\sin (a_1 - \beta_1)} \text{ db} \qquad (1a)$$

where $(a_1 - \beta_1)$ equals ψ_1.

Likewise, movement AC will be sensed in direction CF and FA, the relation being:

$$\frac{CF}{FA} = 20 \log_{10} \frac{\sin (a_1 + \beta_2)}{\sin (a_2 - \beta_2)} \text{ db} \qquad (1b)$$

where $(a_2 - \beta_2)$ equals ψ_2.

2.2. *Orthogonal, asymmetric orientation*

Suppose the sensing movements take place in an orthogonal way (still asymmetric to the normal), and further

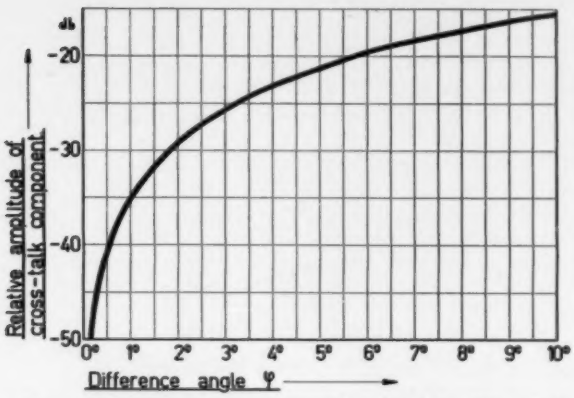


FIG. 5. Relative crosstalk levels as a function of the difference in axes orientation.

assume that cutting has taken place along completely different-orientated moving axes. In trigonometric terms, this condition is reached when in Equations (1a) or (1b), $(\alpha_1 + \alpha_2) = 90°$. The crosstalk equation then reduces to:

$$20 \log_{10} \cot \psi_{1 \text{ or } 2}.$$

This form has been drawn graphically in Fig. 5. It may be observed that even such small angles as 1.8° will lead to relative cross modulation levels of − 30 db.

2.3. *Slanted and rotated plane of axes*

In the preceding paragraphs it has been presumed that the plane of axes is always perpendicular to both disc surface and groove walls. This may not necessarily be true, since much depends upon the way the cutter head or playback cartridge is mounted.

For simplicity's sake, the calculations are based upon the presumption that cutting takes place in a *slanted* and *rotated* plane of axes, sensing being done in a perfectly perpendicular plane. A perspective view of the slanted and rotated array is given in Fig. 6.

Cutting takes place in the plane of ABE along the direction of AB and AE. With respect to the sensing plane of AB′E′ this cutting plane is slanted γ degrees and rotated around AC δ degrees. The lines AB′ and AE′ are in fact projections of AB and AE respectively on the plane of sensing. In viewing the figure, it should be borne in mind that AC is the bisector of angle BAE, β being half the cutting angle; furthermore B′E′ is *not* perpendicular to AC′. Consequently AB′ does not equal AE′ (AB′ > AE′). The position of plane ABE may perhaps be best visualized as the wings of an aeroplane, banking in a turn.

The relation between α and angles β, γ and δ may be expressed in:

$$\cos \alpha = \frac{\sin^2 \epsilon + \cos^2 \beta \cos^2 \gamma - \sin^2 \beta \cos^2 \delta}{2 \sin \epsilon \cos \beta \cos \gamma} \qquad (2)$$

in which ϵ is related to β, γ and δ as follows:

$$\sin \epsilon = \sqrt{1 - (\sin \gamma \cos \beta - \sin \beta \sin \delta)^2}$$

This can hardly be called a simple relationship; to simplify them, assume the cutting angle to be 90°, hence $\beta = 45°$.

Equation (2) reduces to:

$$\cos \alpha = \frac{\cos \gamma + \operatorname{tg} \gamma \sin \delta}{\sqrt{2 - (\sin \gamma - \sin \delta)^2}} \qquad (3)$$

where ϵ has been eliminated. From the denominator it becomes clear that $\cos \alpha$ attains a minimum value, hence α a maximum, when γ equals δ. Figure 7 gives Equation (3) in graphical form; the varying difference angle ψ $(= 45° - \alpha)$ plotted against the angle of slant γ, angle of rotation δ being the parameter.

It is also clear that only for absurdly large angles of rotation and slant will the difference angle ψ be large enough to become important.

THE INFLUENCE OF CROSSTALK ON THE STEREO SOUNDPICTURE

Basis-distortion

The apparent width of the stereo soundpicture is defined as the *basis*. In the playback process, this basis can of course not be greater than the spacing between the two loudspeakers.

Any crosstalk between the two channels will lead to modification of the original basis. *Adding* a cross modulation component R/p to the signal L will cause the sound source L to move toward the R-source. Hence, identical, *in-phase crosstalk components* L/p and R/p will make the sound basis appear smaller. Reciprocally, *out-of-phase crosstalk* will cause *increase* of the original basis (Fig. 8).

As already pointed out, increase of basis is limited to the effective measured distance between the centers of the loud-

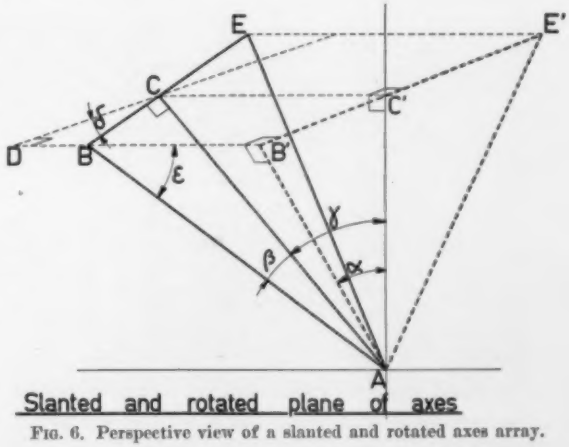


FIG. 6. Perspective view of a slanted and rotated axes array.

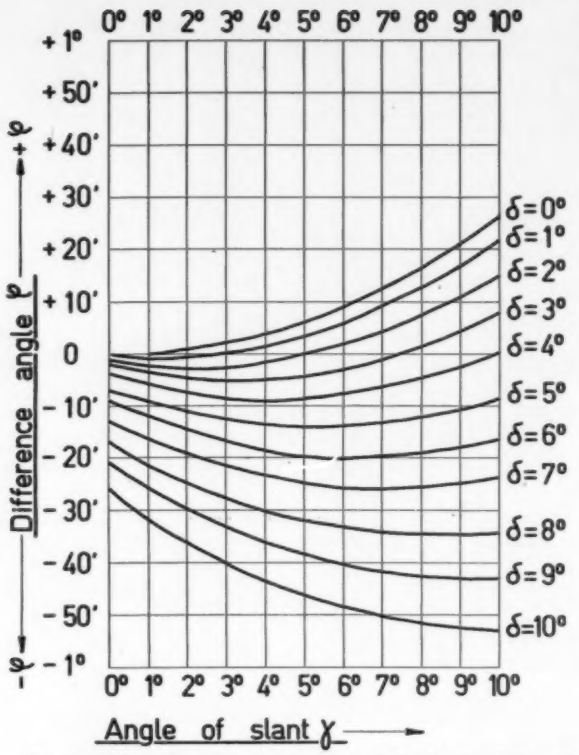


FIG. 7. Difference angle as a function of slant and rotation of the plane of axes.

speakers. Increase beyond this limit will cause the localization of sound sources to disappear.

If for some reason or another, one crosstalk component is in phase and the other out of phase (L + R/p and R − L/p), the sound picture would be decreased in basis in the L-direction and increased in the R-direction (Fig. 9).

This effect will be called *panoramic basis-shift*.

Panorama-distortion

When both signals L and R bear the same information and are equal in amplitude, the apparent sound source would be localized exactly midway between the loudspeakers, provided overall translation characteristics of either channel are equal. Any deviation in amplitude relationship will cause the sound source to move off center, either to the left or to the right, depending on which speaker emits the loudest signal.

Directional hearing as a function of intensity ratios between two loudspeakers has been investigated by K. de Boer.[1] The intensity ratio N at the ears is plotted against the listening-angle θ (Fig. 10).

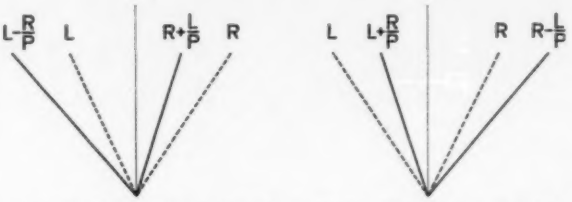


FIG. 9. Depending on the phase relationship between crosstalk components, lefthand or righthand shift of panorama may result.

From the graph it can be seen that the intensity ratio in *decibels* varies in *linear* relationship to angle θ for values of $\theta < 40°$.

Assume the listener to be positioned on the mid-perpendicular to the junction line between the speakers, to wit a distance from the center equal to the distance between speakers (Fig. 11).

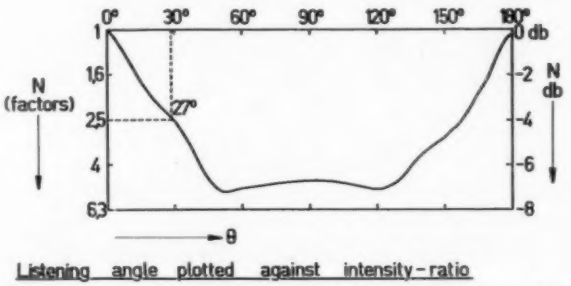


FIG. 10. Directional hearing as a function of sound-intensity ratios.

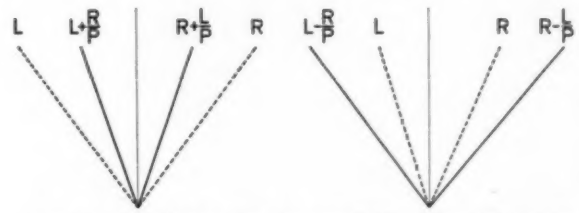


FIG. 8. Depending on the phase condition of crosstalk components, the original basis may appear smaller or larger.

This listening condition is considered to be a practical one. The listening angle between mid-perpendicular and either speaker axis will then amount to about 27°. From Fig. 10 it follows that the intensity ratio N_1 corresponding to this particular value of θ is 4 db.

The linear relationship between N (db) and θ allows the next equations to be made:

[1] K. de Boer, "Stereophonic Sound Reproduction", *Philips Tech. Rev.* (April, 1940).

$$N = \left(4 \times \frac{\theta^\circ}{27^\circ} \right) \text{ db (for any value of } \theta \text{ up to about } 40^\circ)$$

and with N in factors:

$$\log N = \frac{4}{10} \cdot \frac{\theta^\circ}{27^\circ} \rightarrow \theta = (2.5 \log N) \times 27^\circ. \qquad (4)$$

Presuming now that the signal intensities at speakers L and R are respectively 1 and p, the resulting intensity ratio at the listener's ears is determined by:

$$N = \frac{p + \frac{1}{N_1}}{1 + \frac{p}{N_1}} = \frac{1 + pN_1}{p + N_1}. \qquad (5)$$

Equation (5) substituted in Equation (4) leads to:

$$\theta = \left(2.5 \log \frac{1 + pN_1}{p + N_1} \right) \times 27^\circ.$$

This is a general form and for the particular case of Fig. 11, where $N_1 = 2.5$, becomes:

$$\theta = \left(2.5 \log \frac{1 + 2.5p}{p + 2.5} \right) \times 27^\circ.$$

With this formula the angular shift of the stereo sound-picture, caused by amplitude distortion of the original signal ratios, may be determined.

SURVEY OF INFLUENCES ON THE SOUND PICTURE

For the convenience of the reader of this report, a survey of soundpicture distortions as caused by several kinds of axis orientation is given in Table I.

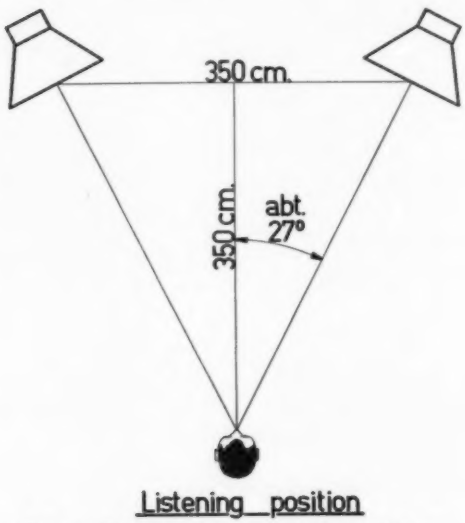
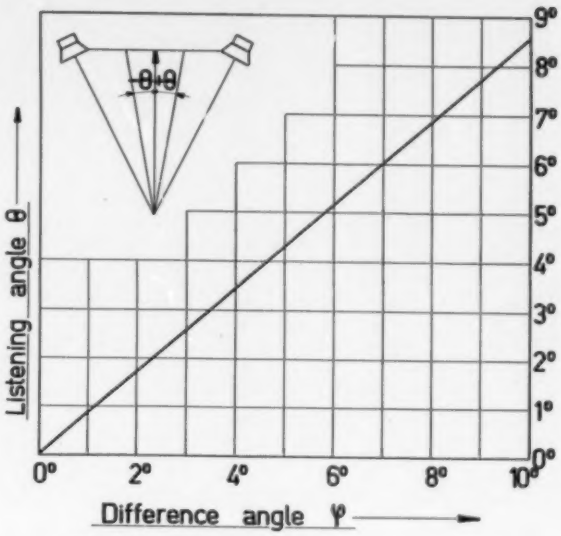


FIG. 11. Plan view of typical listening condition.



When $\psi_1 = \psi_2$, θ is zero.
When $\psi_1 = -\psi_2$, θ is negative (i.e., anti-clockwise rotation).
When $-\psi_1 = \psi_2$, θ is positive (i.e., clockwise rotation).

FIG. 12. Practical values of differences in orientation of axes, plotted against the resulting variation of listening angle.

Calculations for the power ratio of L- and R-channels depending on the difference-angles ψ_1 and ψ_2, are based on the fact that the included modulation angle $\beta_1 + \beta_2$ is always supposed to be 90°.

A general equation is formed in:

$$p_{\frac{L}{R}} = \left(\frac{\cos \psi_1 - \sin \psi_1}{\cos \psi_2 - \sin \psi_2} \right)^2 = \frac{1 - \sin 2\psi_1}{1 - \sin 2\psi_2} \qquad (6)$$

where $p_{\frac{L}{R}}$ = power ratio of L- to R-channels, $\psi_1 = \alpha_1 - \beta_1$ and $\psi_2 = \alpha_2 - \beta_2$. Note that the sine-terms of Equation (6) may cause change of sign, when ψ_1 or ψ_2 is negative.

With $\psi_1 = \psi_2$ (*i.e.*, equal in value and sign), power ratio will be unity and hence *only (symmetrical) basis decrease or increase* will take place and *no change in panorama.*

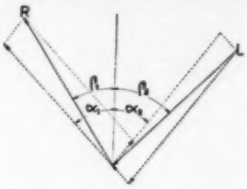
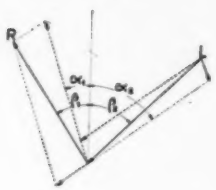
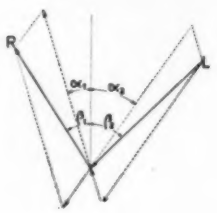
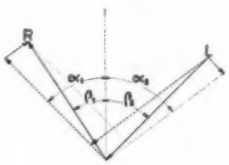
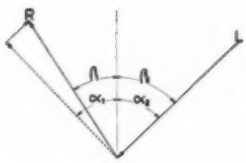
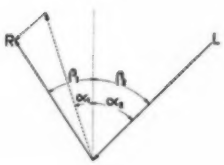
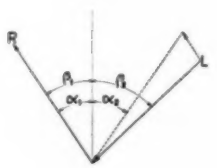
With $\psi_1 = -\psi_2$ (*i.e.*, equal in value and opposite in sign), power ratio will be determined by:

$$p_{\frac{L}{R}} = \frac{1 + \sin 2\psi}{1 - \sin 2\psi}.$$

Finally, for equal angles ψ_1 and ψ_2, Fig. 12 gives the variation of listening Angle θ plotted against ψ_1 and ψ_2.

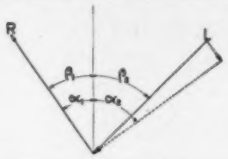
As a practical example it may be noted that for an angle

TABLE I. Survey of Sound Picture Distortion as Caused by Several Kinds of Axis Orientation.

Axis orientation (——— = modulation axes, ····· = sensing axes)	Angles	Interphase relationship of crosstalk	Phase relationship of cross modulation and L–signal	R–signal	Basis distortion	Panorama distortion
	$\alpha_1 > \beta_1$ $\alpha_2 < \beta_2$	anti phase	anti phase ($\psi_1 =$ pos)	in phase ($\psi_2 =$ neg)	panoramic basis— shift to the *left*	
	$\alpha_1 < \beta_1$ $\alpha_2 > \beta_2$	anti phase	in phase ($\psi_1 =$ neg)	anti phase ($\psi_2 =$ pos)	panoramic basis— shift to the *right*	
	$\alpha_1 < \beta_1$ $\alpha_2 < \beta_2$	in phase	in phase ($\psi_1 =$ neg)	in phase ($\psi_2 =$ neg)	total basis— *decrease*	
	$\alpha_1 > \beta_1$ $\alpha_2 > \beta_2$	in phase	anti phase ($\psi_1 =$ pos)	anti phase ($\psi_2 =$ pos)	total basis— *increase*	
	$\alpha_1 > \beta_1$ $\alpha_2 = \beta_2$	—	anti phase ($\psi_1 =$ pos)	— ($\psi_2 = 0$)	Lefthand basis— *increase*	Angular shift to the *left*
	$\alpha_1 < \beta_1$ $\alpha_2 = \beta_2$	—	in phase ($\psi_1 =$ neg)	— ($\psi_2 = 0$)	Lefthand basis— *decrease*	Angular shift to the *right*
	$\alpha_1 = \beta_1$ $\alpha_2 < \beta_2$	—	— ($\psi_1 = 0$)	in phase ($\psi_2 =$ neg)	Righthand basis— *decrease*	Angular shift to the *left*

When $\psi_1 \vee \psi_2$—the angular shift is directed to the right.
When $\psi_1 \wedge \psi_2$—the angular shift is directed to the left.
When $\psi_1 \equiv \psi_2$—no panorama distortion takes place.
N.B.: $\psi_1 \equiv \alpha_1 - \beta_1$ and may therefore have *negative* values.
$\psi_2 \equiv \alpha_2 - \beta_2$

TABLE I (continued).

	Angles	Interphase relationship of crosstalk	Phase relationship of cross modulation and L–signal	R–signal	Basis distortion	Panorama distortion
	$\alpha_1 = \beta_1$ $\alpha_2 > \beta_2$	—	— $(\psi_1 = 0)$	anti phase $(\psi_2 = \text{pos})$	Righthand basis— *increase*	Angular shift to the *right*

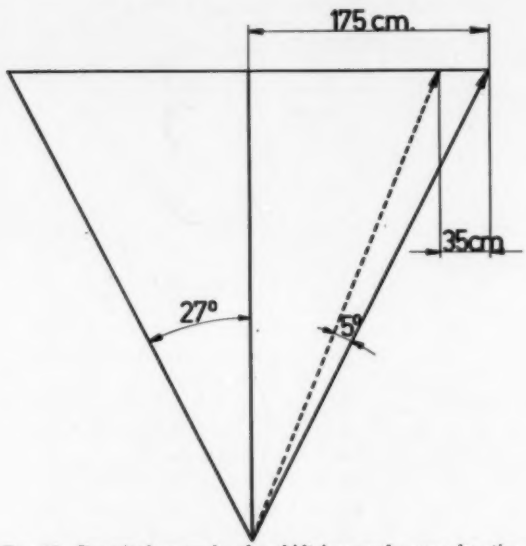


FIG. 13. Practical example of a shift in sound source location as caused by crosstalk.

$\theta = 5°$, the linear basis decrease may amount to some 35 centimeters in either extremity of the covered soundpicture (Fig. 13).

CONCLUSIONS

Since the attenuation of crosstalk products obviously depends to a great extent upon the degree of exactness to which the sensing axes of a stereo pickup cartridge coincide with the modulation axes of a stereo cutting head, it is advisable in the case of the standard 45/45 system to keep the included angle between axes close to 90°, the bisector being normal to the disc surface.

Slant and/or rotation of planes of axes have negligible effects on crosstalk. Consequently, playing of stereo records with tone arms exhibiting substantial tracking errors, or playing on changing mechanisms where the height between turntable and needlepoint may vary depending on the number of records on the turntable, need not lead to deterioration of inherent crosstalk attenuation.

Crosstalk in stereo reproduction leads to various sound picture distortion, such as basis decrease or increase, in which latter case localization becomes hardly possible.

The Deformation of Plastics with Hard, Spherical Indentors*

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The response of plastics to stress when deformed by contact with hard objects under load is important in such diverse applications as bearings, guides, and phonograph records. A study of the permanent deformation of Plexiglas by spherical indentors under a wide range of load, indentor diameter, and time of loading has shown that a single relationship can be used to relate these and other variables. Preliminary sliding experiments at extremely low speeds (ca. 10^{-5} cm/sec) indicate that the relationship derived for static indentations can be used to predict the permanent deformation to be expected in sliding under conditions of diverse loads and speeds. A particularly valuable application is that of predicting the deformation properties of proposed phonograph record systems by comparison with the known properties of existing systems.

INTRODUCTION

THE GROWING USE of plastics in the fields of bearings, guides, and general construction requires a knowledge of how these materials react under stress. In particular, the nature of response (including time-dependence) when such materials are deformed by contact with hard objects under load is especially important.

A familiar example of plastic use, and one where more study is needed, is in the common phonograph record. As can be shown by brief analysis, it is surprising that such a record can be used at all for more than a few playings without receiving destructive damage. Currently, a modern phonograph record (12-inch, 33⅓-rpm) is pressed with a 90° groove (100 grooves to the inch) running spirally in from an outer diameter of 11.5 inches to an inner diameter of 4.75 inches. A hard stylus (generally diamond or sapphire) with a spherical tip (nominal diameter of 0.005 cm) is made to slide at speeds varying from 50-20 cm/sec^{-1} along this groove. Information is stored on the record by undulating the groove so that the center line of the groove corresponds to an electrical signal to be picked up by a mechanical transducer attached to the stylus. The peak-to-peak amplitude of the undulation varies from vanishingly small values for soft, high-frequency notes to a maximum of about 0.0025 cm for a loud, low-frequency note.

The dimensions of the record groove are so minute that any deformation by the phonograph needle would be thought to be excessive. Yet deformation under the applied load must occur or else the loading pressure would be infinite. It has been argued that the groove wall does deform elastically under load, but relaxes after some elapsed time to regain almost its original form. Another difficulty arises, however, in that the elastic yield limit of these materials, as usually calculated, is considerably exceeded by the 0.005-cm stylus loaded at the quite low force of 5 gm. The answer ordinarily given to this is that the force must be applied for some finite time before plastic deformation can occur, this time being longer than the present contact time of the stylus on the record.

Unfortunately, the experimental evidence supporting these statements is meager. Most experiments have been designed to further the art of record reproduction and have shed little light on the mechanism of deformation in the record material. A typical experiment, for example, involves the measurement of the progressive signal loss of a high-frequency note after successive playings at various loadings.[1]

The importance of visco-elastic effects has been demonstrated and discussed by Max.[2] He showed that record wear was minimized when styli of recommended size and loading were used only if sufficient time were allowed between playings for the record material to recover. He further showed that immediately repetitive playings grossly accelerated wear on polystyrene and, presumably, on vinylite. The conclusion was that the successive playings deformed the plastic while it was still in a state of strain and there-

* Received September 18, 1958.
† Chemistry Research Department.

[1] F. Langford-Smith, *Radiotron Designer's Handbook*, pp. 712-717, (Sydney, Australia, Amalgamated Wirless Valve Co. Pty., Ltd., 1954).
[2] A. M. Max, *J. Audio Eng. Soc.*, 3, 66 (1955).

TABLE I. Effect of Load on Indentation Size.

Load (g)	Indentation diameter, d	
	$D = 0.005$ cm, $t = 15$ sec	$D = 0.16$ cm, $t = 15$ sec
5	1.17×10^{-3} cm	
7	1.54	
10	1.96	
15	2.42	
20	2.79	
30	3.27	
40	3.75	
60	4.48	
80	5.18	
100	5.67	
9,080		60×10^{-3} cm
13,620		73
18,160		79

TABLE II. Effect of Loading Time on Indentation Size.

Time of loading (sec)	Indentation diameter, d		
	$D = 0.005$ cm, $W = 20$ g	$D = 0.015$ cm, $W = 180$ g	$D = 0.16$ cm, $W = 13{,}620$ g
1	2.38×10^{-3} cm	6.70×10^{-3} cm	
3			69.7×10^{-3} cm
5	2.50	7.10	70.2
10	2.63	7.70	71.7
20			73.4
25	2.60	8.00	
50	2.75	8.45	76.1
100	2.90	8.55	79.2
200	2.90	8.85	
500	3.00	9.40	

fore were able to effect permanent plastic deformation.

Hunt[3] proposes another factor to be considered. The yield strength of many materials is higher for minute specimens than for bulk samples; hence, the region around a small indentor is protected by the "size effect" against plastic yielding. This "micro-hardness" theory is disputed by Barlow,[4] who says that the apparent increase in hardness is due to elastic springback of the material.

In the light of these questions, it appeared to the authors that it would be well to know whether existing data on the plastic-elastic deformation of plastics could be used to predict their behavior in systems of diverse loads, contact times, and indentor diameters. In some preliminary sliding experiments, we discovered that a variety of effects were operative. Initially, a standard phonograph needle was moved slowly (about 0.1 cm/sec) along the surface of a transparent vinylite record. Simultaneously, we observed the track from underneath through a Metropolitan-Vickers microscope. Vinylite was observed to flow rather like warm butter under the influence of the needle. The track left behind the needle bore no resemblance to the original surface. Obviously, this behavior was quite different from what must occur at the faster sliding speeds used on phonographs.

Observations were then made on deformations during sliding at different speeds. In this work, the plastic, in contact with a diamond stylus under known force, was mounted on a spring-loaded instrument and then released. The spring constant was chosen so as to give sliding speeds starting at zero and accelerating to 350 cm/sec^{-1}. The tracks formed in this manner were found to be dependent, both in appearance and in width, on the material used as well as the sliding speeds. Cellulose acetate, for example, showed transitions from plastic deformation to what appeared to be brittle fracture during the course of changing speed. On the other hand, Plexiglas and cellulose (in the smooth track portion) gave rise to smooth-walled tracks, but showed interesting small scallops in the bottom of the tracks. For Plexiglas, these scallops were formed at the rate of 250,000 sec^{-1} independent of the actual sliding speeds. These observations indicated that the phenomena are complex and, in particular, quite time dependent. It appeared therefore that meaningful interpretations could be made only if the deformation behavior of such materials were studied over a wide range of loading times.

For simplicity, we decided to focus our attention on just one aspect of this process; namely, the extent of permanent indentation resulting from deformation of plastic materials under static loads. Here we were concerned with testing the applicability and validity of empirical expressions relating the time of loading, the size of the indentor, the size of the indentation, and the load applied to the indentor.

EXPERIMENTAL PROCEDURES

The experimental apparatus for the 0.005-cm and the 0.015-cm indentors consisted of an analytical balance in which one balance pan was fitted with a diamond stylus at the bottom of the pan attached by means of a friction clamp. The styli used were conventional GE 0.001-in. and 0.003-in. diamond phonograph needles. These needles were used in their supplied mountings, less the magnetic cartridge assembly. The radii listed were accepted as correct. By means of this arrangement, loads of a few milligrams to 500 gm could be applied to the indentor in a smooth and controlled manner.

The measurements on the 0.160-cm steel ball were made with similar apparatus. The plastic was placed on the pan of a 50-lb. beam balance. The ball, held in a magnetic cup, was pressed against the surface of the plastic just hard enough to balance the beam at the particular setting and held in this position for the time specified.

The indentation diameters were measured on a calibrated Metropolitan-Vickers microscope. The microscope was focused for maximum contrast on surface detail, and the diameter of the indentation was measured with a reticule eyepiece. The measurements were limited both by the difficulty of locating the smaller indentations in the field of view and

[3] F. V. Hunt, *J. Audio Eng. Soc.*, 3, 2 (1955); *J. Appl. Phys.* 26, 850 (1955).

[4] D. A. Barlow, *J. Audio Eng. Soc.*, 4, 116, (1956).

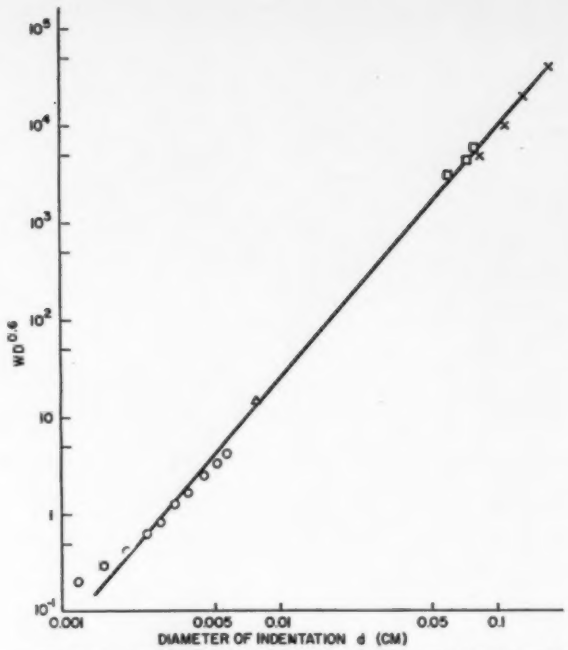

FIG. 1. The effect of load and indentor diameter on indentations in Plexiglas. (Loading time = 15 sec). OD = 0.005 cm; △, D = 0.015 cm; □, D = 0.16 cm; X, D = 1.0 cm (Pascoe and Tabor). W = load in gm. Straight line slope = 2.6.

of determining the actual extent of the indentation for the larger and more shallow indentations.

RESULTS AND DISCUSSION

The experimental results are given in Tables I and II. Each measurement is the average of at least three separate determinations in which the individual measurements varied by not more than ± 10 percent. Further, the indentation measurements were rechecked by a different observer to offset the cumulative effects of eyestrain. It should be remarked, however, that certain consistent inaccuracies can occur because of the difficulty of maintaining a uniform standard of definition over the wide range of diameters and penetrations studied.

A study of static indentations has been carried out for Perspex and other plastics by Pascoe and Tabor.[5] They were able to correlate results on a series of materials using various loads and loading times. However, they used rather large indentors (1.0-cm) only, so it was questionable whether their arguments could be extended to the much smaller size range. A discussion of the principles of static indentations is given in their paper. They point out that indentations made for different-sized indentors can be correlated directly only if the ratio of indentation diameter (d) to indentor diameter (D) is the same for the different indentors; that is, the indentations must have the same geometry except for a scale factor relating the diameter ratios. This follows directly from Myers' law, $W = Kd^m$. This law states that if an indentor of diameter D, under load W forms an indentation of diameter d, then the indentations having the same d/D support the same mean pressure $W/4\pi d^2$. If then, we follow the example of Tabor and assume a power relation, we can write

$$W/d^2 = K(d/D)^x \tag{1}$$

The quantity x is related to an empirical quantity m such that

$$W = Kd^m/D^{m-2} \tag{2}$$

or

$$WD^{m-2} = Kd^m \tag{3}$$

For most plastics, m has been shown to fall between the limits 2.5-2.9.[5]

Pascoe and Tabor chose to correlate their results by plotting $WD^{0.6}$ *vs* log d, which according to the above relationship should give a straight line of slope 2.6 for Perspex. This graphing procedure appears to test only the internal consistency of the data and does not afford a sensitive test of the accuracy of the data or a reliable measure of the index m. Other nearby values of m yield lines of about equivalent internal consistency. However, for comparison, we have plotted our data as well as those of Pascoe and Tabor on a similar basis, as shown in Fig. 1. It is apparent that the results fall on a single straight line of predicted slope except for a slight, consistent deviation of our data in the range of the smallest indentations. It is not possible at this time to state whether this constitutes a verification of Hunt's microhardness proposal[3] in view of the possibility of a consistent experimental error.

In addition to the effect of load and indentor size on the indentation, there is also the effect of loading time. Pascoe and Tabor stated that the indentation area ($\pi d^2/4$) should increase linearly with the logarithm of time of loading and obtained a straight line on such a plot. In the absence of more explicit evidence, there is no reason why such a relationship should be expected to hold, although certainly the indentation size will vary as some function of time. It seems more desirable to combine the effects of time of loading with the other variables into one equation. This can be done by assuming that the proportionality constant K in Eq. (1) is a function of time and that the original equation only holds for constant loading times. The more general equation is then

$$W/d^2 = \left(\frac{d}{D}\right)^x K(t). \tag{4}$$

Now a plot of WD^{m-2}/d^m *vs* $f(t)$ should correlate all

Continued on page 128

[5] M. W. Pascoe and D. Tabor, *Proc. Roy. Soc.* (London) **235A**, 210 (1956).

High Speed Tape Duplication Up-To-Date*

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PLAYBACK SPEED AND RETAIL PRICE

EVER SINCE the advent of practical magnetic tape recording some ten or twelve years ago, there has existed a continuous pressure on manufacturers of tape and tape recorders to improve performance and reduce the cost.

A tape speed of 30 ips was originally required to achieve the desired performance characteristics. It was soon shown that the same performance, or very nearly the same performance, could be achieved at half that speed. Continuing economic pressure reduced the speed once again a few years later by a factor of two, without seriously jeopardizing the overall performance for many users.

Then came stereo recording and the necessity for putting twice as much information on the tape as had previously been done. The result was the 7½ ips two-track stereo tape which made its first commercial appearance in 1955. The retail price of such an item varied from roughly $7.00 to nearly $20.00, depending on the length of the recording. This price structure placed a serious handicap on the widespread sale of stereo tape recordings when compared to the monophonic LP disc which has been available at $3.95 to $5.95. True, the stereo performance from tape yielded a new dimension to home listeners and gave them a medium which did not deteriorate from repeated playings. Furthermore, the signal quality from the tape was felt by many to be superior to that from the LP disc.

That the high price of commercial tape releases limited the market has always been obvious. It was only natural that considerable thought should have been given to reducing this cost. The largest single item in the cost of a tape recording is the raw tape itself. The obvious answer was to increase the packing density of the recorded material. The next logical steps were to cut the speed in half once again, making it 3¾ ips; and to increase the number of tracks from two to four. If one pair of tracks runs in the reverse direction, we have the added advantage of ending the total playing cycle with a rewound tape.

All other things being equal, when the tape speed is reduced by a factor of two we can also expect to cut the frequency response by a factor of two. If we reduce the track width by a factor of two it is axiomatic that we decrease the signal-to-noise ratio by 3 db.

Everybody knows that the 3¾ ips speed was put on tape recorders merely to conserve tape and make it possible for the old man to practice his speeches at home before he went out to give lectures, as I am doing now. Everybody knows that a restricted band width, when applied to speech, has no deleterious effects . . . but everybody also knows that to do the same thing to music makes it completely unacceptable, since the cymbal crashes sound like two loaves of bread being slammed together. Everybody knows that 3¾ ips musical tapes are just not worth listening to. Is this so? It may have been up till recently.

DUPLICATION FOR SLOW-SPEED PLAYBACK

One important link in the entire system which is necessary to bring commercial music to the public, from the original recording thereof to the final playback of a tape in the home, is the high speed tape duplicator. If we accept the fact that a 7½ ips tape can yield a 15,000 cps top end, then we also know that a tape at 3¾ ips can yield only a 7,500-cycle top end; or can it? This used to be true . . . but not any more. The reason for this is that up until a short time ago all playback heads were optimized for 7½ ips or faster speeds. 3¾ ips speed just went along for the ride and the results were seriously handicapped. If a playback head is designed specifically for operation at 3¾ ips it *is* possible, on a production basis, to obtain 15,000 cps with the same uniformity of frequency response that we previously obtained at 7½ ips. There are some other difficulties connected with this situation, however, but we will touch on these a little later.

With the recent introduction of the new commercial tape speed of 3¾ ips, a whole new horizon in the recording industry was opened. This horizon, around its entire perimeter, with everything in between, was just full of problems. We had to go to work again. Our biggest limitation at 3¾ ips is the saturation of the tape itself at signal levels. Since signal-to-noise ratio would thus become the limiting condition in the making of the new 3¾ ips tapes, it was obvi-

* Presented at the Sixth Annual Western Convention of the Audio Engineering Society, Los Angeles, California, February 17, 1959.

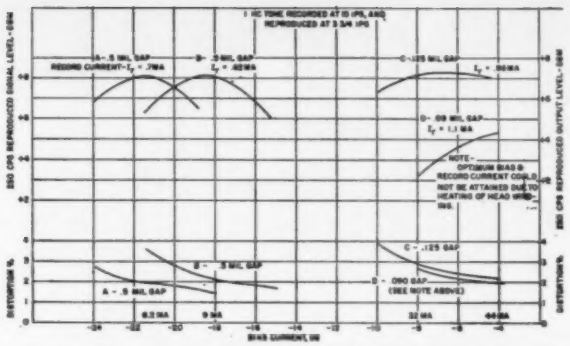

FIG. 1. Bias current *vs* output level and distortion for four duplicator record heads.

ous that we should endeavor to optimize the entire system, starting from the original recording.

By means of optimizing the equalization in the original recording, we were able to improve the signal-to-noise ratio in the master. By using four wide tracks on ½-in. wide tape for the duplicator master we were able to maintain virtually all of this signal-to-noise improvement. This duplicator master is a second generation. Commercially sold copies, or third generation, on ¼-in. wide tape, are made directly from this duplicator master. The track width on the ¼-in. tape is .043 in.

We mentioned the fact that an optimized playback head was necessary to do justice to the 3¾ ips tape speed. Do similar problems confront us when we are discussing recording heads?

In tape duplicating, it is well known that we run the tape at many times its normal playing speed, and make several copies at once, in order to reduce the labor cost. On the Model 3200 duplicator, which is used for 7½ ips stereo tapes, the record head gap is 500 μ inches in length. The tape travels at 60 ips. The band pass of the system is 120 kc. In the first such duplicators which were sold, the head resonance was placed above 120 kc, for obvious reasons. The result was that the signal level recorded on the tape was found to be insufficiently high, and the signal-to-noise ratio was correspondingly poor. An increase in the number of turns on the head increased the signal level from the tape but also brought the resonance down into the band pass. An acceptable operating point was reached, however, around 85 kc, if appropriate equalization was incorporated in the machines. The signal-to-noise ratio was thus improved by quite a margin.

Although this data held for 7½ ips releases, would it hold for the new 3¾ ips tapes?

RECORD HEAD GAP AND TAPE SPEED

Initially it was thought by some that a shorter gap length for the duplicator record head would improve the situation. We tried it and found some interesting results.

The objectives of the investigation were two: (1) to determine whether the narrow gap head would afford 3¾ ips duplicated recordings having improved frequency response, greater signal-to-noise ratio and less distortion; and (2) to ascertain whether or not narrow gap record heads afforded advantages which might be compromised by the proposed 3¾ ips recording characteristics.

The original record head gap length we furnished was 500 μ inches. Since this was felt by some to be too long a gap, we have checked duplicator record heads having 500, 125 and 90 μ inch gap lengths. To test these we have measured the frequency response, distortion, saturation, record level, bias requirement, and signal-to-noise ratio of the recordings made on the high speed duplicator. Measurements were made at a playback speed of 3¾ ips, since this would be the criterion for performance.

To date we have used a 30 ips duplicator master playback speed and a 15 ips slave recorder speed with a pass band top limit of about 65 kc. The 500 μ inch gap head has been wound for a resonance of around 70 kc, since it was found desirable to keep the resonance out of the pass band for reasons of the equalization effects on the signal-to-noise ratio for these 3¾ ips releases.

The performance of two experimental short gap duplicator record heads was compared with that of two types of 500 μ inch record heads, "A" and "B". Head "A" was the prototype of the production version, and head "B" was an experimental model of slightly different construction but with essentially similar characteristics. All of the heads were constructed of the same metal, were laminated, and were .043 in. wide, plus or minus .0005 in.

Each experimental head was mounted in a similar manner. The azimuth and track position were adjustable, and the tape path geometry was similar to that of the standard head. It should further be noted that when speaking of a head we really mean stereo pair, which records tracks 1 and 3 with a spacing between the tracks of .093 in. Various means were used to obtain the gap lengths as stated. The methods used to obtain these precise lengths (which, incidentally, are repeatable) are beyond the scope of this paper.

Standard Ampex Model 3300 duplicator record amplifier electronics and bias oscillator were employed in a conventional manner. The record and bias currents were measured at the jacks of the switch panel immediately adjacent to the head. Test measurements of head current were also made at the record head terminals themselves and found to correspond to the measurements made at the test jacks.

The tapes used for all tests were selected for uniformity of bias requirement, output, and frequency response. The same reel of tape was used for all tests of each head, but different tapes were used for the respective heads. The tape used was considered to be representative of the average of the normally available thin based tape (1800 feet on a 7-in. reel).

The data to be presented are all shown on a comparative basis for clarity.

Figure 1: In each case the bias was adjusted for optimum

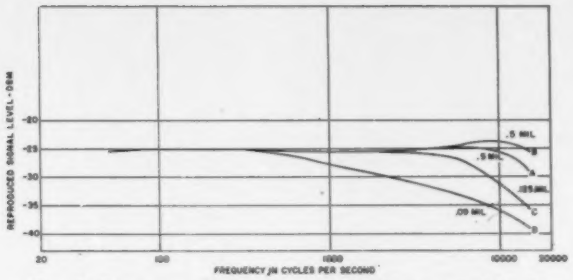

FIG. 2. Frequency response for four duplicator record heads.
Recorded at 15 ips
Reproduced at 3¾ ips
Bias peaked at 1 kc (250 cys at 3¾ ips)
Record pre-emphasis constant.

response so that a 1 kc signal recorded at 15 ips (the duplicator slave speed) would afford a maximum signal level at 250 c when the tape was reproduced at 3¾ ips. In other words optimum bias was established. In Fig. 1 the peak bias requirements and the shape of the biasing curve near the peak are shown for each of the four heads. The minimum bias current is to the left. This figure shows that the prototype head "A" with a 500 μ inch gap requires the least bias, experimental head "B" with the 500 μ inch gap requires a little higher bias, the 125 μ inch head requires a considerable amount of bias. It also shows that it was not possible to obtain an optimum bias point for the 90 μ inch head, because of undue overheating of the head windings themselves. Also shown on the slide are the corresponding distortion levels plotted against bias current for each of the four heads. The distortion decreases as the bias current is increased, but, of course, the high-frequency signal level drops also as the bias is increased above the optimum point.

Figure 2: The frequency responses of the recordings made by each of these four heads having gaps ranging from 500 μ inches to 90 μ inches are shown in Fig. 2. These curves are the result of recording on the 15 ips duplicator

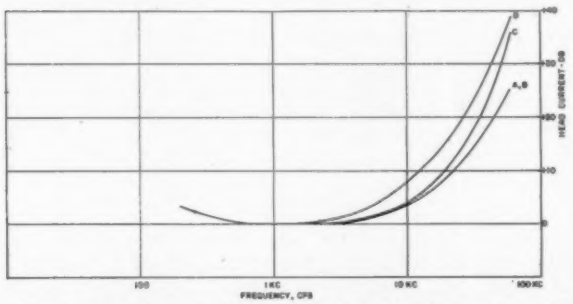

FIG. 3. Record pre-emphasis at 15 ips for four duplicator record heads to yield uniform response at 3¾ ips.
A. Pre-emphasis for 500 mil head
B. Pre-emphasis for 350 mil head
C. Pre-emphasis for 125 mil head
D. Pre-emphasis for 90 mil head

for reproduction at 3¾ ips, and were measured, of course, at 3¾ ips. The bias was peaked at 1 kc at 15 ips as per the data in the last figure, and the recording current pre-emphasis was the same in all cases. It is clear that both 500 μ inch heads afford satisfactory, uniform frequency response. The 125 μ inch head shows a very serious loss, and the 90 μ inch head has a dismal response.

Figure 3: If we are to improve the situation shown in the previous figure, it is obviously necessary that we must introduce additional record current pre-emphasis to compensate for the high frequency response losses. Fig. 3 shows these pre-emphasis curves, relating head current to frequency. Curve "A" was used for the 500 μ inch recordings. Curve "B" was that required for the 125 μ inch head, and Curve "C" was the curve required for the 90 μ inch head.

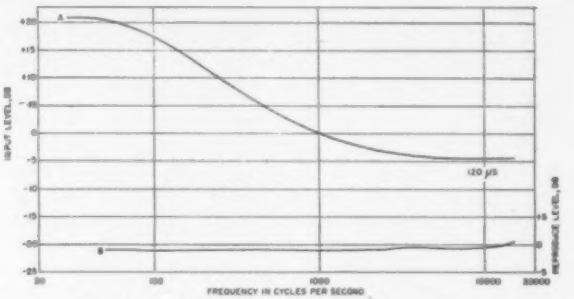

FIG. 4. Playback amplifier response for 3¾ ips.
A. Amplifier response
B. Standard 3¾ ips—4 track tape system response

Figure 4: Although all of these recordings were made on the high speed duplicator, the measurement of the resulting signals on the tape were all made on a modified Ampex Model 351. This machine was modified to play at 3¾ ips with a short gap optimized 2-track playback head and special playback equalization, as shown in curve "A". The playback of a standard tape, corrected for known tape level deviations, is shown in Curve "B".

Figure 5: It will be recalled that the frequency response curves for the shorter and shorter gap record heads resulted in loss of high frequency response. This was corrected by additional pre-emphasis in the recording characteristic. Fig. 5 now shows what happens to the recording when the same level of recording is put on the tape by each of the four heads. The performance of head "A" (500 μ inch gap) is plotted on the top curve, the 90 μ inch head performance is shown by the bottom curve. Six and one-half mils bias was used for head "A", 9 mils for head "B", 32 mils for the 125 μ inch head, and 42 mils for the shortest gap head. The curves show a loss of high-frequency response under these conditions of uniform signal level on the tape, as the gap is shortened. This loss is due to tape saturation. The 500 μ inch head is down 2 db at 12,000 c, the 90 μ inch head is down 2 db at 9,000 c under the same conditions.

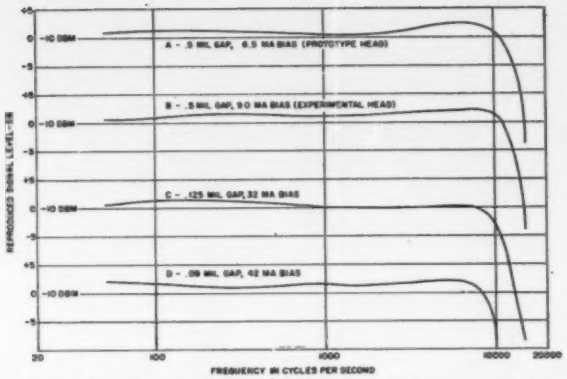
A - .5 MIL GAP, 6.5 MA BIAS (PROTOTYPE HEAD)
B - .5 MIL GAP, 9.0 MA BIAS (EXPERIMENTAL HEAD)
C - .125 MIL GAP, 32 MA BIAS
D - .09 MIL GAP, 42 MA BIAS
REPRODUCED SIGNAL LEVEL-DB
FREQUENCY IN CYCLES PER SECOND

FIG. 5. Signal saturation as a function of duplicator record head gap length.

Recorded at 15 ips; at –15 db below operating level
Reproduced at 3¾ ips
Record current pre-emphasis as per Fig. 3.

CONCLUSION

It has been established that optimizing the playback head for 3¾ ips will yield the playback results we wish. Adequate high-frequency response requires a short straight gap with very sharp edges to yield adequate resolution of these short wavelengths recorded on the tape.

On the other hand, it may now be indicated without much question that a short gap *record* head does not necessarily yield superior results, at least as far as the high speed duplication of tapes is concerned. We have noted that, if the gap length is reduced, an increase in bias current is required. When we are considering the possibility of using several slaves at once, it is clear that we may get into difficulty with respect to our bias supply source if we wish to use the shorter gap heads. The shorter the gap length, the more pre-emphasis in recording current is required, in order to compensate for the loss in signal level at the higher frequencies. The more pre-emphasis used parallels the increased tendency toward tape saturation at high frequencies. This type of tape overload causes the signal quality to deteriorate rapidly and provides a less desirable musical rendition.

All things considered, it looks very much as though we had picked the right gap length the first time, and that the shorter gap lengths, which some have thought might possibly be better, haven't worked out in actual practice.

The final objective has been to improve this link in the magnetic recording chain, among others, to the point that the production of 3¾ ips four-track stereotapes becomes practical and more closely comparable in price with discs. Although one may always continue to improve a system, this present achievement is a distinct step forward in the art.

THE AUTHOR

Russell J. Tinkham attended the University of Oregon and was graduated from the University of Illinois, where he studied mechanical engineering. He was formerly associated with Armour Research Foundation, with Magnecord as president, and with Shure Brothers. Now manager of audio sales for Ampex Corporation, Mr. Tinkham is also a member of the Acoustical Society of America, the IRE, and the SMPTE.

Continued from page 124

measurements on a given material. It may be seen in Fig. 2 that the diverse experimental points for these results on Rohm and Haas Plexiglas as well as Pascoe and Tabor's results on Perspex do indeed lie on one common line within experimental error. The important thing to note here is that, even for these widely varying sizes of indentors (D = 0.005 – 1.0 cm) and loads (5 – 13,600 gm), a single relationship can be used to correlate the several parameters over a considerable range of loading times (1 – 1000 sec.).

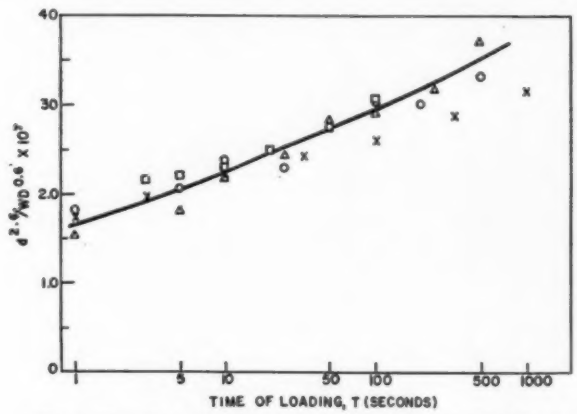

FIG. 2. The effect of loading time on indentations in Plexiglas for various indentor diameters. O,D = 0.005 cm; △, D = 0.015 cm; □, D = 0.16 cm; X, D = 1.0 cm (Pascoe and Tabor). W = load in gm, d = indentation diameter in cm.

Some preliminary sliding experiments at extremely low speeds (ca. 10^{-5} cm/sec) yielded track widths comparable with the static indentation diameters shown in Fig. 2. As discussed by Pascoe and Tabor, the loading time is assumed to be the time necessary for the stylus to travel a distance equal to the track width. This observation indicates the validity of combining static and sliding information into a single relationship. Further experiments at increased sliding speeds are necessary to establish the validity of Eq. (4) for contact times of less than one second.

The relationship expressed by Eq. (4) can now be used, for example, to predict the extent of permanent deformation caused by diverse combinations of styli diameter, load, and sliding speed in phonograph systems. It is only necessary to assume that the single relationship shown in Fig. 2 extends to the much lower contact times (ca. 10^{-5} sec.). Since the actual position of the curve is still unknown for these lower times, it is necessary to make the prediction by comparison with known systems.

A New Device for the Reduction of Print-Through*

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The *Echoraser* is a static device designed to reduce the intensity of layer-to-layer signal transfer or print-through. Each set consists of two *Echorasers*, one "Red" and the other "Black". The "Red" device will remove up to 9 db, and the "Black" device up to 18 db, of print-through.

The *Echoraser* can rejuvenate old original recordings damaged by print-through and can virtually remove all the print from new recordings. This device can be a valuable tool for the recordist who wishes to prevent his recordings from developing objectionable print-through.

INTRODUCTION

THE MAGNETIC TAPE industry during the past few years has made considerable progress in combatting print-through, or layer-to-layer signal transfer. The marketing of low print Master Audiotape in 1957 was a giant step in this direction, though not a complete remedy. For years experimenters have demonstrated that print-through can be effectively reduced by applying a slight erasing field to a recorded tape as it passes over a head just before reproduction. An erase head is used, powered by either the bias or the erase current of the recorder. Surprisingly few recordists have availed themselves of this technique, which led us to ponder the reasons for this resistance. Perhaps the thought of tinkering with the internal circuits of the recorder discouraged some, or there was not enough time for experiment to determine the proper combination of head and current to do the job, and finally the possibility of tape damage in the event of component failure may have been a deterrent. Whatever the reason, it was felt that a simple device which could be installed without a knowledge of electronics, without lengthy pre-testing, without wires or terminals, could bring this technique to more recordists. The device called an *Echoraser* is quite unique in construction and yet extremely simple in design. In spite of the simplicity, the reduction in print-through achieved by the *Echoraser* follows the same patterns observed by earlier experimenters using more complex equipment.

DESCRIPTION

The *Echoraser* is designed to produce an "erase" field of pre-determined intensity. Figure 1 is a drawing of the component parts. The main body of the device is 1½ inches in height and the front face is divided into two areas. The smaller area is a bearing surface for the tape to be treated and the larger area is the "active" or energized surface. The base is a thin metal plate ½ inch by ¾ inch, fitted with two positioning pins which slip into holes in the main body. The base plate has two screw holes to allow permanent installation on a tape transport. The energized area of each is colored to denote its erasing intensity.

The energized areas are made by cementing sections of specially prepared magnetic tape to the surface of the device. This tape has been pre-magnetized with a special pattern to create the desired erase intensity when the recorded tape passes by.

To use, the recorded tape is allowed to pass over the unit just before it passes by the playback head of the recorder.

PERFORMANCE

The amount of print-through reduction achieved with the *Echoraser* is dependent on several factors. They are

1. the storage conditions of the recorded tape;
2. the age of the recording;
3. the printing frequency or wavelength;
4. the magnetic properties of the tape.

Storage Conditions

The conditions of storage affect the performance. If the recorded tape has been subjected to high temperatures or exposed to external magnetic fields, the print will be harder to reduce. The results achieved are proportional to the care given to the recordings during storage. Even with tapes exposed to the poorest of storage conditions some reduction in print-through can be realized.

Age of Recording

It is common knowledge that the longer a recorded tape

* Received September 18, 1958. Read before the Tenth Annual Convention of the Audio Engineering Society, New York, October 1, 1958.

† Director of Quality Control

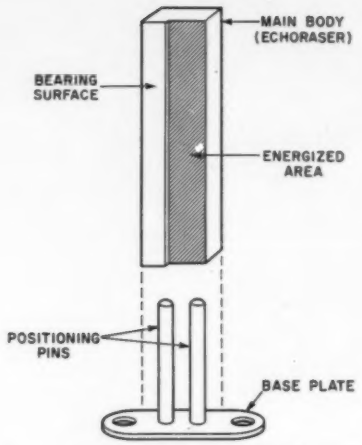

FIG. 1. Component parts of the *Echoraser*.

is stored the more difficult it becomes to erase. A device with fixed erase intensity becomes less effective as the age of the recording increases. To cope with this problem, two models were developed, each with a different power. The red *Echoraser* will remove up to 9 db of print-through without appreciably affecting recorded high-frequencies. The black *Echoraser* will remove up to 18 db of print-through, at a sacrifice of 2 to 3 db of recorded high-frequency level.

Figure 2 shows the reduction of print-through, plotted against the age of a recording made on a low-print Audio Mastertape. The tape was recorded at 15 ips with a 1 kc signal of saturation intensity, at peak bias, on every fourth layer of tape on the reel. The tape was stored at room temperature and tested periodically for two years. As shown in Fig. 2, one hour after recording, the black *Echoraser* removed 18 db of print while the red *Echoraser* removed 9 db. After two years storage of the tape the black unit is still capable of removing 6 db of print, and the red unit removes 3 db of print.

PRINT FREQUENCY

The reduction in print-through varies with the wavelength of the printing signal. A frequency of 1 kc at a tape speed of 15 ips, or a wavelength of 15 mils, is used throughout this paper as a test frequency. This wavelength was selected because it produces the most severe print on the standard tape having a 1.5 mil base tape and a 0.5 mil magnetic coating. Using this wavelength as a reference point, note the variation in print-through reduction with the black unit as the wavelength changes in Fig. 3. At a wavelength of 60 mils or 250 cps, 22 db of print-through is erased. At 2 kc, only 15 db of print-through reduction is realized. The shorter the wavelength of the printing signal, the less the reduction in print-through; it is fortunate that print-through itself is also less at the shorter wavelengths.

MAGNETIC PROPERTIES

The *Echoraser* has been designed for standard tape of domestic manufacture, which ranges from 230 to 300 oersteds in coercive force. Using the *Echoraser* against tapes exceeding 300 oersteds in coercive force will destroy some of its power. If tapes with coercive forces below 230 oersteds are treated, extreme caution should be taken to avoid excessive erasure. In general, the higher the coercive force of the tape, the less will be the print reduction achieved by the *Echoraser*.

The Master tapes used to demonstrate the effectiveness of the *Echoraser* in this paper have a coercive force of 260 oersteds. One may achieve better or poorer results with a specific tape, depending on its magnetic properties.

SIGNAL ERASURE

The amount of signal erasure also varies with the age of the recording. The red unit removes less than 1 db of signal and can be disregarded as a problem. However, the black *Echoraser*, which has at least twice the erase intensity of the red unit, will reduce high-frequency signals from *newly recorded tapes*. Figure 4 shows the loss of 15 kc signal from the black unit plotted against the age of the recording. Again, a Master tape was used with a 15 kc signal recorded at 30 db below 3% harmonic distortion, at peak bias, and at a speed of 7.5 ips. The black *Echoraser* removed 3 db of signal after one hour of aging. The amount of signal removal diminished until no removal was observed after 10

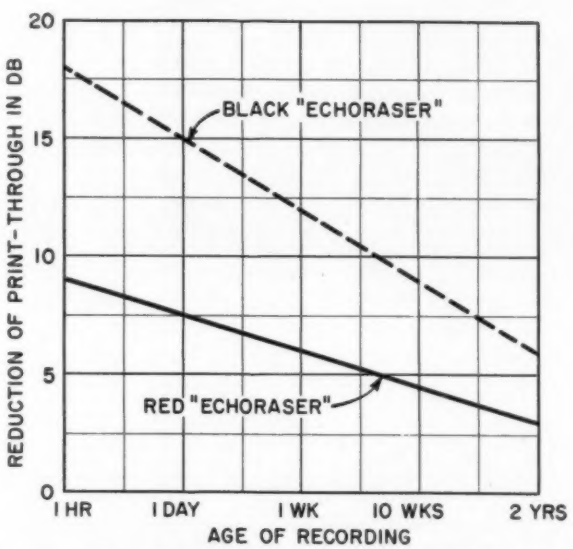

FIG. 2. Print reduction *vs* age of recording. Test made using standard reference recorder, Master Audiotape, speed of 15 ips, signal frequency of 1 kc, at saturation level, with bias for peak output.

weeks of aging. The removal of low-frequency signal was less than one db after one hour of aging and was also disregarded, for this paper.

The black unit should not be used unless the recording being treated is at least ten weeks old, or unless one is prepared to sacrifice some high-frequency response to reduce the print-through. The wise recordist might benefit by recording his future tapes with a slight high-frequency accentuation, thereby anticipating the erasing treatment.

If there is any doubt as to the age of a given recording, the red *Echoraser* should be used to be safe.

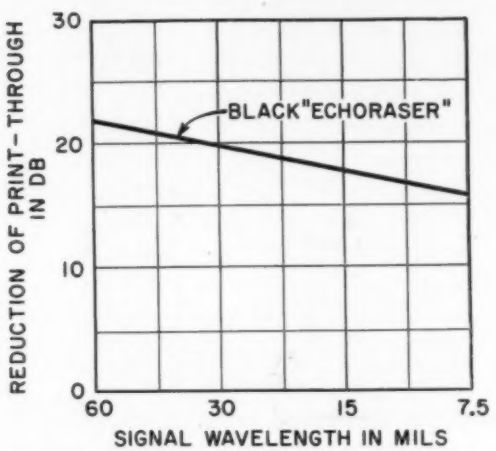

FIG. 3. Print reduction *vs* wavelength. Test made using standard reference recorder, Master Audiotape, speed of 15 ips, saturation recording level, with bias for peak output.

MORE PERFORMANCE DATA

To further illustrate the importance of the *Echoraser* in eliminating print-through as a serious recording problem, Fig. 5 shows the improvement in signal-to-print-through ratio obtainable on a Master tape, plotted against the age of the recording. One hour after recording, the signal-to-print-through ratio is reduced to 82 db with the black *Echoraser* and to 73 db with the red *Echoraser*. After two years the black *Echoraser* restores the signal-to-print-through ratio to 62 db, which is the print ratio displayed by the Master tape after only one day of aging.

These units when used properly exhibit no harmful effects. Exhaustive tests showed no effect on either harmonic or intermodulation distortion. Listening tests revealed no additional noise. The background noise of recorded tapes lessened, resulting in a "cleaner" sound. It is safe to assume that the confused background noise due to overlapping print signals was reduced along with the distinctive prints.

Wear tests equivalent to 500 passes of a reel of tape failed to reduce the efficiency of the device. The units have been designed to compensate for wear.

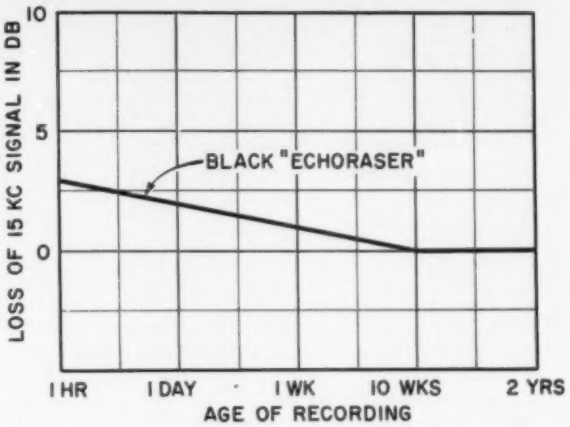

FIG. 4. 15 kc signal loss *vs* age of recording. Test made using standard reference recorder, Master Audiotape, speed of 7½ ips, recording level 30 db below 3% distortion at 1 kc, signal frequency of 15 kc, with bias for peak output.

Repetitive passes past an eraser do not increase the reduction of print-through, as previous experimenters have noted. However, the eraser will take care of print which has developed after the last print erasing, since this is new print.

A print-through eraser works only on the print-through on the original tape. It is useless to rerecord a tape with heavy print onto a second tape and then to use the eraser on the second tape. The reason for this is that on the original tape the print is much easier to erase than the signal; where-

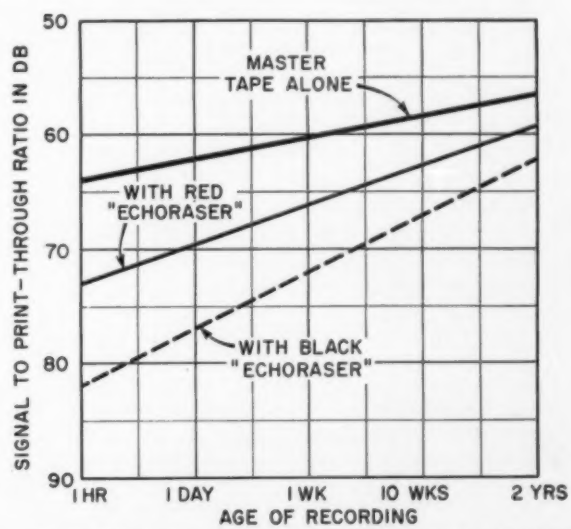

FIG. 5. Effect of age of recording on print-through ratio. Test made using standard reference recorder, Master Audiotape, speed of 15 ips, signal frequency of 1 kc, at saturation level, with bias for peak output.

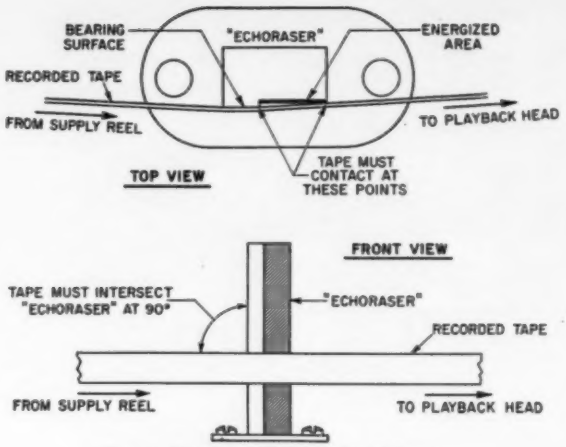

FIG. 6. Placement diagram for *Echoraser*.

as in copying the print is recorded solidly into the second tape by the bias.

INSTALLATION AND MAINTENANCE

The proper placement of the *Echoraser* is essential to achieve the full benefit of print-through reduction. The device must be placed between the supply reel and the playback head, accessible for easy removal and placement. The upper section of Fig. 6 is a top view showing the tape contacting the front face. It is extremely important that the oxide surface of the tape be in contact with both the bearing surface and the edge of the energized area, as indicated by the two arrows, to insure proper functioning of the device. The lower section of Fig. 6 is a front view showing the tape intersecting the front face at an angle of 90°. The intersecting angle should not vary more than plus or minus 5°. On virtually all tape transports the vertical positioning is automatic, since the tape travels parallel to the top plate.

Before permanent installation it is advisable to temporarily affix the device in position with "contact" cement, or the like, and to run a few experimental reels to check proper alignment and operation. Special care must be taken to avoid placing the *Echoraser* too far into the line of travel of the tape. This may cause excessive pressure, resulting in frictional vibration. The tape must contact the *Echoraser* firmly but lightly for ideal operation. Make certain that the tape does not touch the base positioning pins when the main body of the device is removed.

To insure continued good performance, the following precautions should be observed:

DO NOT place *Echorasers* near external magnetic fields.
DO NOT place energized surfaces together or against any metal surfaces.
DO NOT clean *Echoraser* with liquids. Use only dry cloth.
DO NOT place *Echoraser* in position against program material when tape is not in motion.
DO NOT leave *Echoraser* in position when rewinding or fast forward winding.
DO NOT leave *Echoraser* in position when not in use.

In general, treat the *Echorasers* as though they were valuable recordings.

CONCLUSION

The *Echoraser* can be an important tool for the recordist. By proper planning of his recording techniques he can eliminate objectionable print-through. He can rejuvenate his old recordings by reducing print-through and background noise. By employing both Master low-print tapes and *Echorasers* he can store his recordings with the assurance that in years to come his material will be least affected by this inherent property of magnetic tape, print-through.

THE AUTHOR

Frank Radocy was born in Stamford, Connecticut, 1921. In 1939 he was employed by Audio Devices, Inc., and has remained with that company to date, with the exception of four years spent as an Army Air Corps Pilot during World War II. He attended special evening courses at the Wright Technical and Bridgeport Engineering Institutes. In 1946 he was appointed Lacquer Department Manager. In 1949 he was appointed Director of Quality Control.

For the past few years Mr. Radocy has travelled throughout the U. S. and Canada, providing field and sales engineering services for the company. He is a member of the Tape Standards Committee of the MRIA.

Mr. Radocy has presented several papers at A.E.S. and I.R.E. Conventions in recent years. His main interests lie in the development of improved and new magnetic tapes and lacquer discs for the industry. He is shown checking the performance of an *Echoraser*.

Applying the Transistor in a Stereophonic Tape System*

DWIGHT V. JONES†

Semiconductor Products Department, General Electric Company, Syracuse, New York

A complete semiconductor, stereophonic tape playback system applies modern solid state devices to a new concept of music appreciation that is receiving more interest every day. This system consists of a stereophonic tape deck, two tape preamplifiers, two 10-watt amplifiers, two 16-ohm speaker systems, and a common power supply. Transistors offer many advantages in high fidelity circuits, since there is no problem with microphonics or hum pick-up from filaments as we have with tubes. Transistors are inherently low impedance devices and thus offer better matching to magnetic pick-ups and loud speakers, for more efficient power transfer.

PREAMPLIFIERS

THE TWO preamplifiers consist of two identical amplifiers with circuitry as shown in Fig. 1. Both preamplifiers use a common 18-volt battery supply. The circuit of Fig. 1 consists of three direct-coupled transistor stages. The first two stages have a feedback bias arrangement for current stabilization of the two stages. The resistor R2 from the emitter of TR2 provides this dc current feedback to the base of TR1. The output stage is well stabilized with a 5K emitter resistance.

The negative feedback from the collector of TR2 to the emitter of TR1 is frequency selective to compensate for the standard NAB recording characteristic. The preamplifier frequency response from a recorded tape is shown in Fig. 2. The flat response from a standard pre-recorded tape occurs with the tone control (R12) at mid-position or 12KΩ. There is 7 to 8 db of treble boost with the control at 25K maximum position, and approximately 20 db of treble cut with R12 = 0.

The preamplifier output is approximately 2 volts with the input being the maximum 400 cycle recorded level for 2% distortion on the tape (Standard Reference Level). The total harmonic distortion of the preamplifier at this level is under 0.2%. With a preamplifier output of 4 volts at 400 cycles, the total harmonic distortion is still less than 1%.

This preamplifier will accommodate a variety of tape head impedances, for it gives an equalized output for a 2KΩ head at 1Kc and also a 6KΩ head. The input impedance of the preamplifier increases with frequency because of the frequency selective negative feedback to the emitter of TR1. The impedance of the tape head also increases with frequency but is below that of the preamplifier.

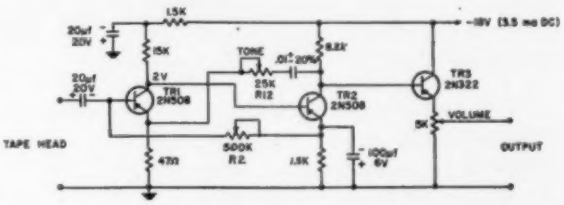

FIG. 1. Tape preamplifier.

The 2N508 was selected for this application because it is basically a high gain, low noise transistor. This transistor operates in a circuit designed to achieve the good signal-to-noise ratio of which the 2N508 is capable. The S/N of this preamplifier is approximately 60 db depending on the tape deck and head that are used. The noise level varies greatly with respect to head structure, shielding, and physical layout.

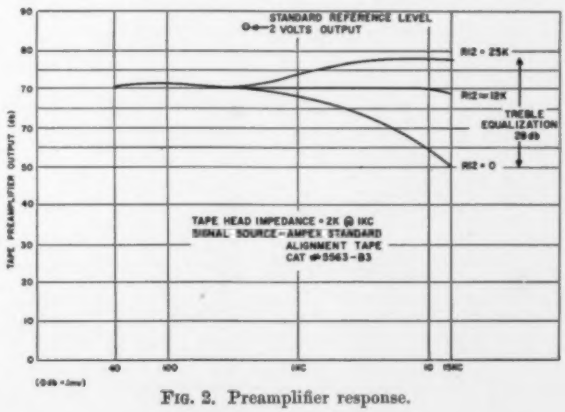

FIG. 2. Preamplifier response.

* Revised copy received June 2, 1959. Delivered before the Tenth Annual Convention of the Audio Engineering Society, New York, September 29, 1958.

† Application Engineer.

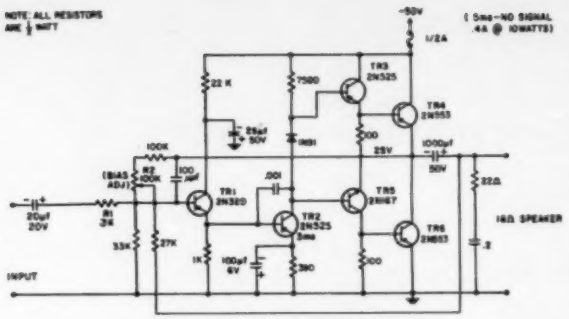

FIG. 3. Power amplifier.

The emitter-follower stage gives a low impedance output for a cable run to the power amplifier, and acts as a buffer so that any preamplifier loading will not affect the equalization characteristic.

POWER AMPLIFIERS

A great deal of effort has gone into developing transformerless push-pull amplifiers using vacuum tubes. Practical circuits, however, use many power tubes in parallel to provide the high currents necessary for direct driving of low impedance loudspeakers.

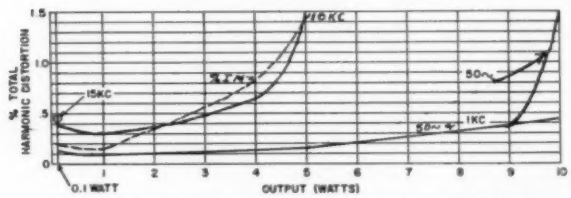

FIG. 4. Distortion *vs* power output for 10 watt amplifier.

The advent of power transistors has given new impetus to the development of transformerless circuits since the transistors are basically low-voltage, high-current devices. The emitter follower stage, in particular, offers the most interesting possibilities since it has low inherent distortion and low output impedance.

The two power amplifiers consist of two identical amplifiers with circuitry as shown in Fig. 3. This is a direct-coupled amplifier with excellent low frequency response, and also has the advantage of a feedback arrangement for current stabilization of all stages. The feedback system also stabilizes the voltage division across the power output transistors TR4 and TR6, which operate in a Class B push-pull arrangement. TR3 and TR5 also operate Class B in the familiar Darlington connection. Using an NPN for TR5 gives the required phase inversion for driving TR6 and also has the advantage of push-pull emitter follower operation. TR4 and TR6 have a small forward bias to minimize cross-over distortion. This bias is set by the voltage drop across the 100-ohm resistors. TR3 and TR5 are biased for the

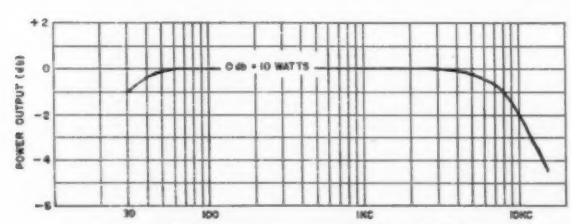

FIG. 5. Maximum power response at clipping.

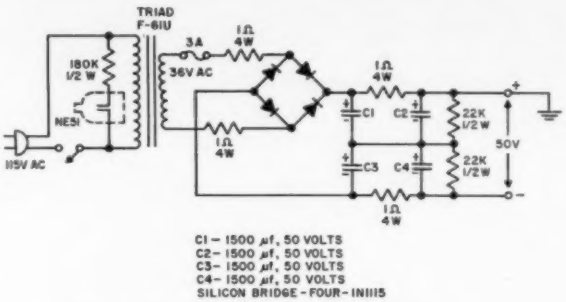

FIG. 6. Power supply for dual 10-watt amplifier.

same reason with the voltage drop across the 1N91. A 68-ohm resistor would serve the same function as the 1N91 except that there would be no temperature compensation. Thermistors have also been used to compensate for the temperature variation of the emitter-base resistance, but they do not track this variation as well as a germanium junction diode, which has temperature characteristics similar to the transistor.

TR2 is a Class A driver requiring a very low impedance drive which is accomplished by an emitter follower TR1. TR1 needs a current source for low distortion; thus R1 was set at the maximum value that would allow the amplifier to be driven to the 10-watt clipping point with the Standard Reference Level into the preamplifier.

The bias adjust R2 is set for one-half the supply voltage across TR6 and can be trimmed for symmetrical clipping at maximum power output. The .001 μf feedback capacitor from collector to base of TR2 aids in stabilizing this circuit by reducing the phase shift and high frequency gain of this stage. The 100 μμf capacitor shunting the bias network further aids the stabilization with high frequency negative feedback from output to input. This circuit has approximately 15 db of overall voltage feedback with the 27K resistor from load to input. The 16-ohm speaker system is shunted by 22 ohms in series with .2 μf to prevent the continued rise of speaker impedance and its accompanying phase shift beyond the audio spectrum.

The overall result, from using direct-coupling, no trans-

Continued on page 140

Automatic Plotting of Cartridge Response

C. P. GERMANO

Clevite Electronic Components, Clevite Corporation, Cleveland, Ohio

A system for the automatic plotting of frequency response of phonograph cartridges is presented. The major parts of the system are a piezoelectric ceramic driver, an RIAA compensating network, and graphic recording equipment. The driver consists of a stack of modified Lead Zirconate Titanate (PZT*) length expanders acting together as a bar. The graphic recording equipment includes a Bruel & Kjaer (B & K) Signal Generator in conjunction with a B & K High Speed Level Recorder. A description of these components is given and typical response curves are shown. The latter are compared with curves obtained in the usual manner.

INTRODUCTION

THE FREQUENCY RESPONSE of phonograph pickups is generally determined by the use of frequency-test records. These are of two general types: banded tone and gliding tone. The former are recorded with a number of constant frequencies which are played in sequence. Gliding tone records are cut with gradually changing frequencies to cover the desired range. Special test records, such as sweep-frequency types, are also available, and they allow visual displays of the pickup response on an oscilloscope. Where graphic plots are desired, however, the aforementioned frequency-test records are required, and the response is obtained in a laborious manner.

It becomes desirable, therefore, to automatically plot frequency response. Specially constructed calibrators have been reported[1,2] in the past, and apparently these have worked satisfactorily. The use of these units has been apparently somewhat limited. The method to be described herein is akin to these. It is felt, however, that some improvements have been effected which have been made possible by recent advances in equipment and transducer materials. Of major importance here is the availability of a piezoelectric ceramic such as PZT with its high electromechanical activity which allows greater displacement for a given applied electric field.

DESCRIPTION OF SYSTEM

The system is made up of three major parts; namely, a piezoelectric driver, a compensating network, and graphic recording equipment with its associated components. Fig. 1 shows a block diagram of the test setup. Initially, it was intended to include the compensating network as part of a feedback loop to control the stack driving voltage in accordance with the RIAA recording characteristic. However, after some experimentation, more suitable results were obtained by driving the stack at constant displacement and compensating the pickup output. The difficulty experienced here was due to the inability to accurately monitor stock displacement for feedback.

The driver unit is formed of a set or stack of modified PZT length expanders which together act in the same manner as a single piezoelectric bar. The unit is excited in a longitudinal direction, but it is oriented at 45° to the vertical for use with stereo phonograph pickups. A small piece of record material is cemented to the free end of the stack to effect mechanical coupling to the stylus. While this does present an additional load to the stack, it is too slight to be of any consequence.

The network shown following the cartridge is a simple RC circuit to compensate for the RIAA recording characteristic. The piezoelectric driver is an amplitude responsive device and yields constant displacement with constant voltage over the useful range.

PZT DRIVER

Before discussing the driver, it may be worth while to say a few words concerning piezoelectricity. Although a full treatment of the subject is not within the scope of this paper, the following remarks will provide sufficient background for an understanding of the action of the driver unit.

Materials which generate electric charges when mechanically excited and conversely become mechanically deformed when electrically excited are known as piezoelectric bodies. Strictly speaking, piezoelectric phenomena are confined to crystalline substances such as quartz, ammonium dihydrogen

* Reg. U. S. Patent Office

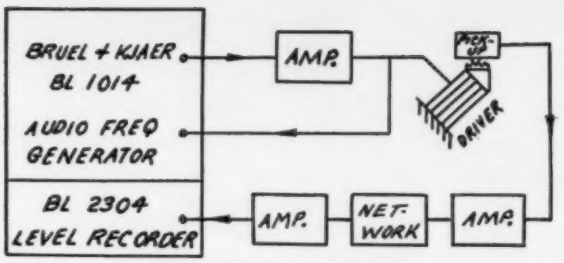

FIG. 1. Block diagram of system.

phosphate, etc. Polycrystalline materials such as barium titanate and PZT in reality exhibit what is known as a polarized electrostrictive effect. However, it has become customary to refer to this effect as a piezoelectric one. These polycrystalline substances do not have piezoelectric properties in their original state. Such effects are induced in the material by the application of strong depolarizing fields.

The electromechanical effect of the polarized material is the development of electric charge opposite to that applied to the electrodes during the polarization when the element is mechanically expanded in the direction of polarization or compressed at right angles to it. Conversely, a signal voltage applied with this same polarity as the original polarization produces further expansion in the direction of the field and contraction in the plane perpendicular to the field.

The coefficients describing expansion or contraction in the same direction as the electric field are given subscripts (*33*). Those describing expansions or contractions perpendicular to the electric field are given subscripts (*31*). The first subscript refers to the direction of the electric field conventionally the *3* or *z* axis. The second subscript refers to the direction of the strain.

The direct piezoelectric effect is referred to as generator action, and the converse effect is referred to as motor action. By this is meant that electrical energy is obtained from mechanical (or acoustical) and vice versa. Thus, piezoelectric elements are, in effect, motor-generators, and as such, their behavior is influenced by the mechanical and electrical loads attached.

Any body possesses certain frequencies of mechanical resonance at which points the amplitude of vibration is relatively large. Similarly, when driven electrically, a piezoelectric transducer vibrates with large amplitudes of displacement at certain frequencies depending upon the size, mode of operation, and type of material. The increase in amplitude at these frequencies over low-frequency operation is dependent upon the mechanical Q of the element and system. Operation below resonance produces a strain or displacement constant with frequency and essentially linear with electric field.

Since it is important to simulate record drive, the displacement to be applied to the cartridge under test is kept constant. Consequently, it is important to construct a driver whose natural resonant frequency is higher than the highest frequency to be covered. Furthermore, it is desirable to obtain displacements of the same magnitude as found in typical records. These two requirements are somewhat in conflict with one another, because the resonant frequency of a length expander (*31*) mode is inversely proportional to the length to thickness ratio of the plate. A compromise becomes necessary. Let us examine the relationships involved. The resonant frequency of a blocked free bar in longitudinal mode is given by

$$f_r = \frac{1}{4l}\left(\frac{Y_{11}^E}{\rho}\right)^{1/2} \tag{1}$$

where Y_{11}^E = Young's Modulus perpendicular to poling axis
ρ = density
l = length of plate

for the material in question this results in

$$f_r \approx \frac{30}{l}\ \text{kc} \tag{2}$$

In order to determine the displacement sensitivity of a length expander, use is made of the piezoelectric d modulus. This is defined as the strain as a function of applied field,

$$\text{that is} \quad d_{31} = \frac{\text{strain}}{\text{field}}.$$

As indicated previously, the 31 subscripts show that the strain is perpendicular to the field.

$$\text{Since strain} = \frac{d}{l}$$

d = displacement

$$\text{and field} = \frac{v}{t}$$

v = applied volts

$$d_{31} = \frac{d/l}{v/t} \tag{3}$$

$$\text{or} \quad \frac{d}{v} = d_{31}\frac{l}{t}\ \text{(displacement per unit volt)}.$$

This expression applies only when operating below resonance and assumes an unloaded element. The latter condition is practically achieved since the mass and compliance of the stylus can be considered negligible compared to the mechanical impedance of the stack.

To keep the natural resonance high, l should be kept small. However, for large displacements l should be made large. It is possible to keep l small, consistent with high resonance, and to increase the displacement by reducing the plate thickness.

Ideally, the resonant frequency should be, perhaps, twice the highest frequency to be covered. However, because of

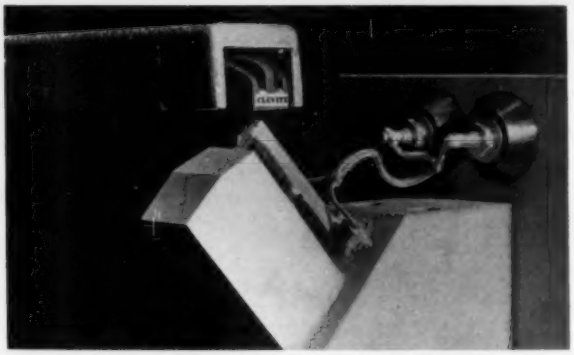

FIG. 2. PZT stack driving a clevite pickup.

the low mechanical Q and the desire to obtain a large displacement, the length of the stack was chosen to correspond to a resonance of 20 kc. As can be seen in Fig. 6, the stack is useable up to a frequency of 15 kc.

This means a length of 1.5 inches and consequently for the material in question,

$$d/v = 0.825\ \mu\text{inches/volt.} \qquad (4)$$

Approximately 350 volts would be required to obtain a displacement of 0.3 milinches which corresponds to a record velocity of approximately 4.8 cm/sec. at 1000 cps.

However, a further compromise is necessary because of the low impedance presented by the stack at the high frequencies. This would necessitate a driving amplifier of high power-handling ability. It was felt that even ⅓ of the above displacement would be adequate to drive the cartridge. For this displacement, approximately 100 volts is required. Fig. 2 shows a closeup of the driver.

COMPENSATING NETWORK

Inasmuch as the ceramic stack is an amplitude responsive device, the displacement sensitivity is constant with frequency up to or near its natural resonant frequency.

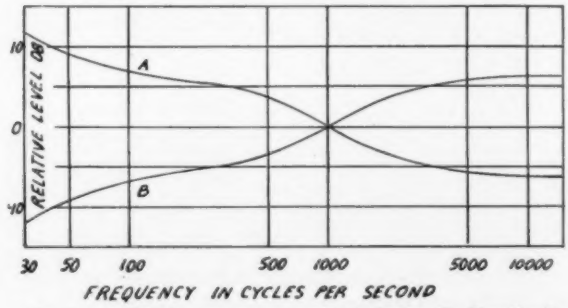

FIG. 3. A. RIAA curve on an amplitude basis. B. Equalization needed by amplitude cartridge.

Consequently, some method of compensation is necessary to correct the cartridge output in accordance with the RIAA recording characteristics.

Curve A of Fig. 3 shows the RIAA characteristic plotted on an amplitude basis. Curve B of the same figure represents the complement or equalization necessary for a constant amplitude cartridge to reproduce such a record with a flat response. Piezoelectric cartridges are normally equalized by the proper choice of load impedance and the judicious use of element and stylus resonance.

Thus, the piezoelectric cartridge is designed to reproduce the RIAA characteristic flat; and since the stack is driven at constant displacement, a network such as is shown in Fig. 4 is necessary to provide the bass boost and treble cut in accordance with Curve A of Fig. 3.

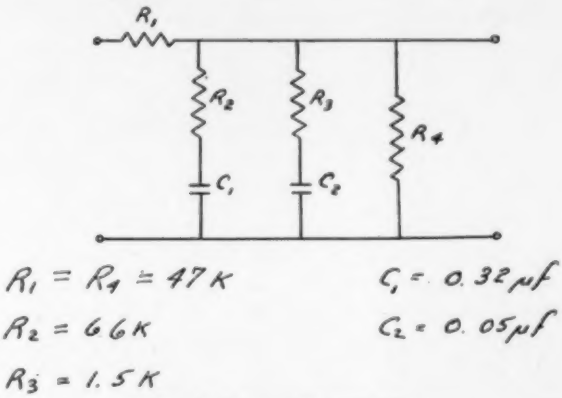

FIG. 4. Compensation network.

There is no intention of going into any detail on the design of this circuit. For information concerning compensation networks, it is suggested that the literature[3] be consulted. Suffice it to say at this point that there are three networks which define the standard RIAA characteristic. Each network provides the proper time constant for three turnover frequencies. These frequencies are at 50, 500, and 2120 cps. The network shown in Fig. 4 provides the proper compensation. This is illustrated in Fig. 5, which shows the response. The dotted line curve of Fig. 5 is Curve A of Fig. 3, superimposed on the network response to show the correlation. Except for the very low end, the two curves match quite well. Fig. 6 shows the displacement sensitivity of the driver. The calibration of the stack was effected by the use of a microdisplacement meter described in literature.[4] Briefly, the principle of operation of this equipment is as follows:

A probe driven by an ADP crystal standard is made to follow the displacement of the test element. The capacity between the probe and element forms part of a frequency-determining circuit which operates at approximately 100 megacycles. With non-synchronous motion between the two,

the oscillation is frequency modulated and this can be detected by a frequency-modulation receiver. When the two are in synchronism, there is no deviation from the center frequency. With this condition, a knowledge of the voltages supplied to both the standard and test elements along with the sensitivity of the former, leads to the displacement of the test element.

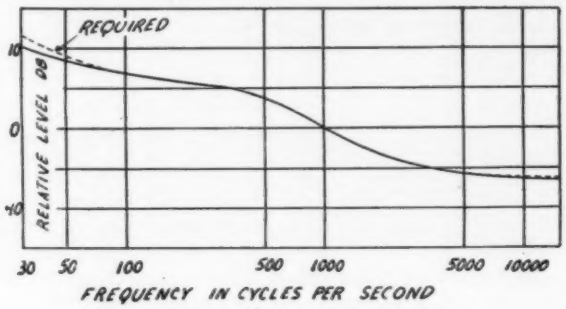

FIG. 5. Response characteristic of network.

GRAPHIC RECORDER

Use is made of a Bruel & Kjaer High Speed Level Recorder in conjunction with a B & K Signal Generator and Control System to obtain the graphic plot of frequency response. The audio-frequency signal generator is a flat-frequency oscillator with provision for automatic frequency scanning and chart recording. The logarithmic frequency range is swept continuously by the level recorder which is mechanically linked to the generator by a flexible drive shaft connection. This provides mechanical scanning of the beat

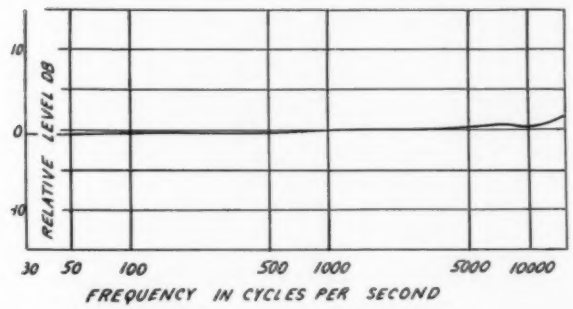

FIG. 6. Displacement characteristic of driver.

frequency oscillator and a synchronization between the chart paper frequency scale and oscillator frequency tuning. The Level Recorder is a peak-indicating ac voltmeter that records level of voltages in the frequency range 20 to 20 kc. Rapid writing speeds allow the precise recording of sudden changes in level of the measured signal, and a range of log-

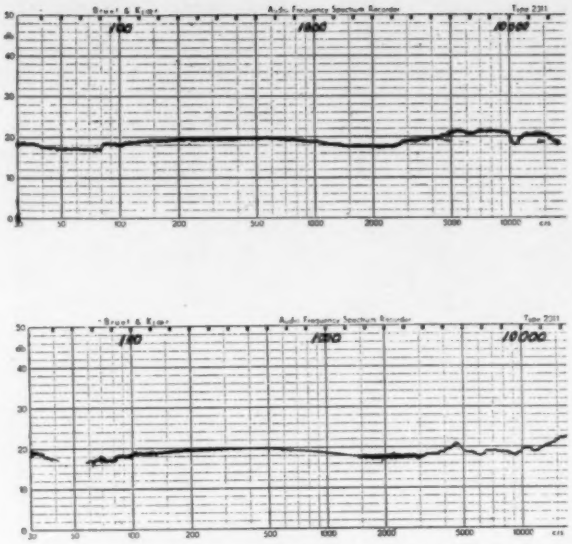

FIG. 7. Frequency response of Clevite experimental stereo pickup (top tracing, right channel; bottom tracing, left channel).

arithmic input circuits permits measurements of great dynamic range.

PERFORMANCE OF SYSTEM

The system described has been used to measure the response of a number of piezoelectric-type stereo cartridges.

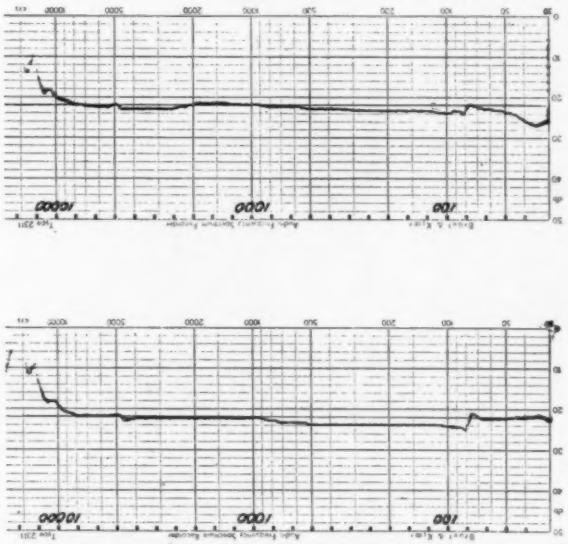

FIG. 8. Frequency response of commercial piezoelectric stereo pickup type A (top tracing, right channel; bottom tracing, left channel).

Typical curves are shown in Figs. 7 through 9. In Fig. 7 is shown the response frequency characteristics of a Clevite experimental model using PZT Multimorphs.† The Multimorph has been previously described,[5] and for a complete treatment of the subject it is suggested that the appropriate reference be examined. This element behaves as a bender or flexure unit, but differs in cross-section in that it is a single-plate unit rather than a composite or two-plate sandwich. A series of parallel holes coated with graphite and running the length of the element provide the central conductor required for polarization to obtain flexure action. Normal benders consist of two expander plates cemented together so that an applied flexural displacement results in additive voltages. When driven electrically, such an element produces bending similar to that of a bimetallic strip. Rather than by unequal expansions, however, bending is caused by expansions of the two plates in opposite directions. For purposes of comparison, Fig. 10 shows the response of the Clevite unit as obtained with the system described, as well as by the use of standard techniques. In the latter case, the Westrex Stereo Disk 1A frequency-test record was used.

This is substantially a constant-velocity recording, and since the cartridge is designed to reproduce the RIAA characteristics, the curve is corrected and replotted showing deviation from flat response. Correlation is seen to be quite good. The discrepancy noted at the high end can probably be attributed to the fact that mechanically the two systems are not exactly alike. The presence of stylus friction in the case of a rotating disk undoubtedly affects the stylus-record resonance.

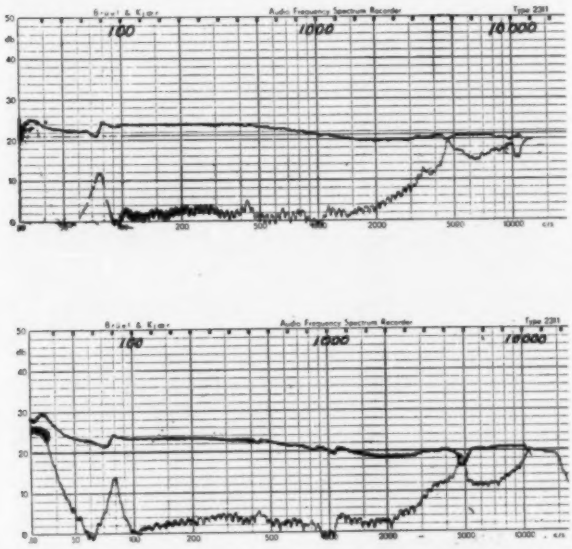

FIG. 9. Frequency response of commercial piezoelectric stereo pickup type B (top tracings, right channel response and separation; bottom tracings, left channel response and separation).

† Reg. U. S. Patent Office

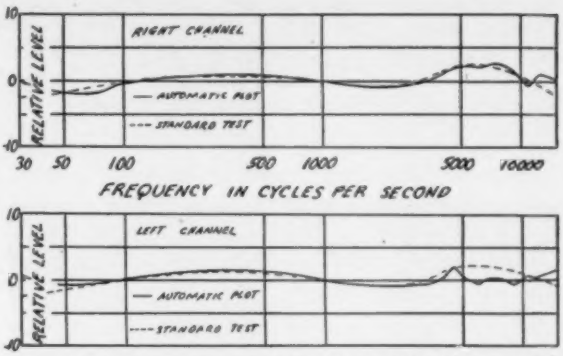

FIG. 10. Frequency response of Clevite experimental stereo pickup.

The response curves shown in Figs. 8 and 9 were obtained on commercially available piezo-electric cartridges. An attempt was also made to measure separation, and this is illustrated in Fig. 9. This curve does not generally correlate well enough with that obtained using standard test records to consider it reliable. As can be seen in Fig. 9, appreciable noise was present.

ACKNOWLEDGMENTS

The writer is indebted to Messrs. E. Kaine and E. Johnson for the construction of the driver and for the performance of the tests. He is also indebted to Mr. E. Maciag for the design of the Clevite cartridge.

THE AUTHOR

C. P. Germano has for many years been interested in piezoelectric elements. His early work was concerned with methods of measurement, and he now designs, develops, and evaluates new or improved crystal and ceramic piezoelectric elements. Previously associated with Victoreen Instruments Company, Mr. Germano joined Clevite in 1950. He was educated at Northwestern and John Carroll Universities, holding a B.S.E.E. and M.S. in physics. He is a member of the Acoustical Society of America, a senior member of IRE, and he holds memberships in a number of other professional societies.

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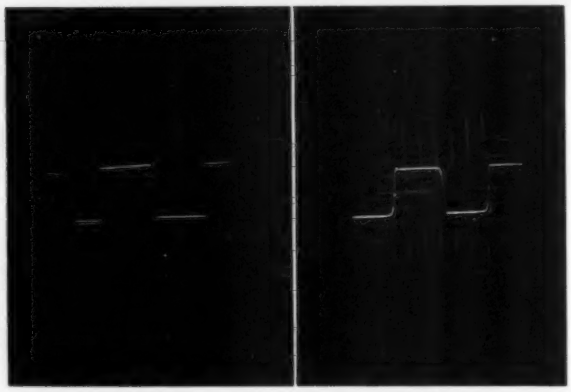

(a) 1 kc (b) 5 kc

FIG. 7. Square-wave responses.

formers, and ample degeneration, is an amplifier with output impedance of one-half ohm for good speaker damping, and very low total harmonic distortion as shown in Fig. 4. The power amplifier S/N is better than 80 db. The frequency response at 100 mw is flat over the audio spectrum. The power response curve is shown in Fig. 5. The amplifier square-wave response is shown in Fig. 7.

The dual amplifier may be operated from a common low impedance power supply as shown in Fig. 6 or from a 30- to 50-volt battery supply.

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Dwight V. Jones is application engineer, Semiconductor Products Department, General Electric Co., Syracuse, N. Y. He was graduated from Kansas State University with B.S.E.E. degree in 1947. Mr. Jones is a member of Eta Kappa Nu and Sigma Tau honorary societies. He was nominated to *Who's Who in Engineering* in 1958 and is a senior member of IRE.

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AES News

In a booth at the Western Audio Engineers Exhibit, Harry Bryant listens to a nineteenth century magnetic wire recorder. This instrument, which is still operable, was invented in 1898 by the "Danish Edison," Valdemar Poulsen. Two U. S. companies manufactured Poulsen's "Telegraphone," but were unable to compete successfully against the lower priced Victrola.

AES News

Glenn Wallichs (right) receiving an AES Certificate of Appreciation from Harry Bryant at West Coast Convention.

THE SIXTH ANNUAL WESTERN CONVENTION

Fifty technical papers were delivered at the highly successful Sixth Annual Western Convention in Los Angeles last February. As was true of the Fall Convention in New York, not only the number but the quality and interest of the papers set a record.

On Monday evening before the convention and exhibition actually began, the Awards Banquet was held at the Biltmore Hotel. Members of the Institute of High Fidelity Manufacturers, whose show coincided with the AES Convention, were guests at the banquet.

Guest speaker at the banquet was Glenn Wallichs, a co-founder and the president of Capitol Records, Inc. Mr. Wallichs received a certificate of appreciation for "his outstanding leadership and encouragement of continuing research in high fidelity engineering."

An AES Fellowship Award went to James A. Miller of Santa Barbara in recognition of his fifty years of activity in the audio field. With many inventions for sound recording and reproduction to his credit, Mr. Miller is still an active researcher. At present he is working on improvements in microgroove transcriptions.

Convention sessions were not confined to technical subjects. An unusual feature was the series of talks called "Stereo for Salesmen." The purpose was to make available to salesmen the new techniques developed by AES engineers.

As Harry Bryant said after the convention, "The audio art received the biggest showing it ever had on the West Coast, thanks to our convention. More than thirty companies and organizations were represented. Society members and audio enthusiasts, by listening to the papers presented at the technical sessions and by examining the equipment in the exhibition booths, had an opportunity to learn the latest developments in audio from the professionals in the field."

AES President Don Plunkett, in a recent newsletter, gave a fitting tribute: "The Sixth Annual West Coast Convention, managed by Harry L. Bryant, was most successful. The entire Society is grateful to Harry and his Committee. Congratulations!"

Joseph N. Benjamin, President of the Institute of High Fidelity Manufacturers, addressing the guests at the AES Convention Banquet.

James A. Miller (right) receiving an AES Fellowship Award from Harry Bryant at the West Coast Convention.

NEW JOURNAL AES EDITOR

At a meeting of the Board of Governors on June 15, Walter H. Erikson (RCA, Camden, N. J.) was invited to become Editor-in-Chief of the JOURNAL. Mr. Erikson accepted the offer and will assume his new duties beginning with the October, 1959, issue.

SECTION NOTES

A significant increase in Section activities reflects the growing vitality of AES and accompanies its expanding membership. The Pittsburgh Section demonstrates what can be done.

They held regularly scheduled and very well attended monthly meetings from October, 1958, through June, 1959. The intimacy and informality of these Section meetings allows for a wide range of discussion topics and social activities, all geared to a continuing interest in audio development. Pittsburgh, for example, opened its 1958-59 season with a talk on "The Science of Sound" given by Prof. Charles Williams of the Carnegie Institute Department of Physics. The season ended in June with an "open house" recording session.

Other Sections were similarly engaged. To give some idea of the extent of Section activities, the following newsletter is reproduced in its entirety.

SUBJECT: *AES Section Newsletter #2*
TO: *AES Sections and Board of Governors*
FROM: *Donald J. Plunkett, AES President*
DATE: *April 15, 1959*

Unfortunately in the last newsletter we did not have explicit information on the activities of our Japan Section. However we have just received reports on the activities of the Section for the past nine months. Information on two 1958 meetings indicates a July meeting with guest speaker, Mr. Reedy, President of Interstate Electronics Corp. The meeting was held at International House of Japan. Mr. Reedy addressed the membership on new developments in electronics in the United States, particularly in the field of stereo and transistor preamps. In October, Mr. Ralph Endersby of the Ampex Corp. addressed a meeting held during the 5th Audio Show, Tokyo, on Magnetic Tape Recording. Both members and non-members turned out to make this one of the most successful meetings of the year for the Japan Section. A January meeting was concerned primarily with organizational problems. Presiding at the meeting were Mr. John H. Nakamura and Mr. Warren Birkenhead. Mr. Birkenhead is from the United States and has been in Japan for several years. He was one of the leaders in the formation of the Japan Section. The February meeting was concerned with Stereo Reproduction in the Home, which seems to be a world-wide topic. (If there are too many conflicting opinions it may have to be referred to the UN. But who would veto it now?) Seriously, however, it would also be interesting to know the Russian Hi-Fi and stereo situation. In March, a description of a four-track stereo disc was made and proved to be the highlight of the demonstration and talk. It was not reported whether or not this 4-track tape machine cartridge was an American or Japanese product. It was most interesting to have these reports on the activities of the Japan Section and I hope their activities expand and that they keep us informed of their many worthwhile meetings.

A January meeting of the Pittsburgh Section was privileged to present Dr. Aubrey Epstein of the University of Pittsburgh on "Hearing and Noise." The meeting was highly technical, covering many medical aspects of hearing, a timely warning on the damage that can result from prolonged listening to high amplitude signals. (I wouldn't reveal this information to your wives, or you will never hear the end of "I told you so.") Secretary McKee also reports that a joint meeting with the IRE was in the formation stages. In March, Mr. Robert Strome of the Ampex Corp. addressed the Pittsburgh Section on "A Short Review of Improvement of Signal-to-Noise Ratio Obtained from a New Tape Record/Reproduce Curve." The Section also voted to protest to CBS over the cancellation of E. Power Biggs' Radio Program.

The Southern Tier (N. Y.) Section has been actively serving its members. In January, a meeting presented Mr. A. Watson of the G.E. Co. in a discussion of Electrostatic Speakers. In March, Mr. Dirk Ross of McIntosh Lab described a Stereo Preamp (the McIntosh C-20). Both meetings were well attended. Business was also conducted during the March meeting with the Chairman reporting on the boundaries of the Section.

The Central New York Section, neighbors of the Southern Tier Section, held a meeting in January presenting Mr. A. C. Angus of the G.E. Co. in a talk on a transistorized audio console. There was a dinner for the membership preceding the actual meeting which proved very popular. (This is a technique used by other sections and always seems to pull a larger audience. Perhaps it might be useful to other sections.)

In March, the San Francisco Section presented a review of papers presented in February at the West Coast Convention held in Los Angeles. Participating in the program were Messrs. Russ Tinkham, John McKnight, Robert Acker and Vincent Salmon.

The 11th Annual Convention of the AES will be held in New York, at the Hotel New Yorker, October 5 to 9. The Chairman of the Convention is Mr. Harry Bryant. This year Mr. Bryant has energetically assumed the operation of both Conventions. It is a tremendous job to pull a convention together, and Harry needs the assistance of everyone. If any member of your Section wishes to present a paper at the Convention this year, please have him contact Mr. Harry Bryant, Radio Recorders, 7000 Santa Monica Boulevard, Hollywood 38, California. This year we again intend to present a Professional Exhibit in conjunction with the Technical Papers presentation. If any electronic manufacturer in the field of audio in your Section area is interested in participating in this year's Exhibit, have him contact AES Exhibit Planning Committee, P. O. Box 12, Old Chelsea Station, New York 11, N. Y. Please look around your area and see if you can provide the Committee with information on any potential exhibitors at the AES Professional Exhibit.

Please keep the meeting reports flowing in order to allow us to circulate your meeting and section activities to our fellow Section Chairmen and members . . . and, of course, keep those meetings going.

Regards,

DONALD PLUNKETT, *President, AES*

Report of the AES Technical Committee on Magnetic Recording

This, the second report of the Committee, repeats the announcement of the formation of the AES Technical Committee on Magnetic Recording. The primary function of this Committee is to keep the membership informed, through the pages of the JOURNAL, of the latest developments in the field of magnetic recording applications to audio engineering.

This Committee can provide this useful service to members of the Society only if those working in the field of magnetic recording contribute material on new developments to the Committee for publication in the JOURNAL. The Chairman of the Committee will personally contact many individuals working in the field of magnetic recording in an effort to reach those who may not have an opportunity to read the JOURNAL. If you have any thoughts relative to the future activities of the Committee, please send them to the Chairman at the address given at the end of this report.

The length of an item that is to be included as part of the Committee's report probably should not be more than one page of the JOURNAL, equivalent to about 1000 words. At the other extreme, the item might consist simply of a photograph or a graph with a single-line caption. The use of appropriate illustrations is encouraged, provided that they are furnished in the form of glossy

photographic prints, ink drawings, or photostat copies of ink drawings. Each contributed item published as part of the report will include the contributor's name and company affiliation. In addition to the new development items, we hope to publish useful bits of engineering information on magnetic recording extracted from notebooks and reports that have not been published. An example of material of this type is as follows:

DISTORTION CHARACTERISTIC OF NICKEL COBALT PLATING

Several years ago a commonly used magnetic recording wire consisted of a nickel cobalt plating on a phosphor-bronze wire base. The plating of 20 percent nickel and 80 percent cobalt was 0.0003 in. thick, and the phosphor-bronze wire base was 0.004 in. in diameter.

These nickel cobalt platings exhibit a most unusual characteristic of third harmonic distortion as a function of high frequency bias current as shown in Fig. 1. The wire speed was 24 ips and the test signal frequency 400 cps. The distortion curve in Fig. 1 was obtained by measuring the third harmonic distortion with a wave analyzer for a large number of values of bias current. At each value of bias current the audio current was adjusted to produce a playback output level 12 db below the saturated output level. Therefore, the curve in Fig. 1 is third harmonic distortion as a function of bias current for a constant playback output level. A similar distortion curve, obtained with a conventional tape head on a nickel cobalt plated drum, indicates that this distortion curve is characteristic of nickel cobalt platings independent of the shape of the recording medium or type of head used.

Chairman of the AES Technical Committee on Magnetic Recording

WALTER H. ERIKSON
Radio Corporation of America
Building 10-8
Camden 2, New Jersey

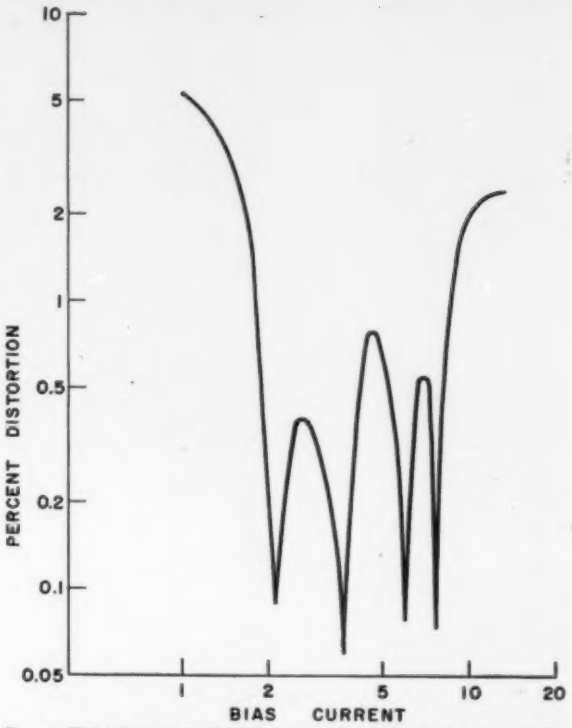

FIG. 1. Third harmonic distortion as a function of bias current for nickel cobalt plated magnetic recording wire.

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Its ease of operation has led to its use by many manufacturers to replace the standard procedure of checking audio amplifiers with an oscillator and a VTVM. Engineers have found very accurate correlation between the two methods, and prefer the intermodulation method because of savings in time and money. Price of the Model 31 is only $220.00. For further information write **Measurements, Boonton, N. J.**

Membership Information

New members and Change of Grade

(April 1-June 8, 1959)

New Members

Members

Abiko, Kenshiro—Tokyo, Japan
Bishop, Richard J.—Binghamton, New York
Cahn, Bernard L.—San Francisco, California
Colby, Chester H.—Groton, Connecticut
Collins, Edward T.—Rockville Centre, New York
Heard, Donald E.—Burbank, California
Heaton, James S.—San Mateo, California
Holtz, Robert F.—Zurich, Switzerland
Howard, Bruce—Fort Worth, Texas
Isberg, R. A.—Menlo Park, California
Korkes, Henry—Brooklyn, New York
Mattern, Murle F.—Menlo Park, California
Morgulis, Isaac A.—Toronto, Ontario, Canada
Naatz, Raymond C.—Burbank, California
Oehding, Heinz—Toronto, Ontario, Canada
Perfetti, Gianfranco—Milano, Italy
Peters, Anthony—Kew Garden Hills, New York
Ross, David—Brooklyn, New York
Sparks, Richard W.—Las Vegas, Nevada
Wilson, W. Paul, Jr.—Sepulveda, California

Associate Members

Allego, Anthony B.—Ambridge, Pennsylvania
Auth, Dale W.—Los Angeles, California
Berman, Louis M.—La Mesa, California
Bucci, J. Richard—Redwood City, California
Dolby, Leman W., III—Pittsburgh, Pennsylvania
Graham, Carl O.—Syracuse, New York
Grimshaw, Richard F.—Syracuse, New York
Hanford, Eugene B.—Oakland, California
Hatfield, Stanley R.—Los Angeles, California
Hayes, Thomas B.—Liverpool, New York
Holdom, John D.—Burnaby, British Columbia, Canada
Jackson, Louis M.—Los Angeles, California
Layton, Robert D.—San Jacinto, California
Lutzick, Daniel—Los Angeles, California
Marco, Romaine L.—San Bruno, California
Matthys, Robert J.—Minneapolis, Minnesota
Moore, Donald O.—Monterey Park, California
Nai, Peter—New York, New York
Nangle, Wilfrid O.—Oklahoma City, Oklahoma
Quinn, Harold E.—Brooklyn, New York
Riecke, Charles L., IV—Johns Island, South Carolina
Ruch, Donald A.—Azusa, California
Schuder, Donald A.—Woodland, California
Sinder, Alfred M.—Dayton, Ohio
Sterman, Arthur—Allison Park, Pennsylvania
Szafir, Emil V.—Austin, Texas
Viti, Alessandro—New York, New York
Webster, Perry C.—Covina, California
Wells, Ralph R.—Van Nuys, California

Students

Anderson, Robert L.—Baton Rouge, Louisiana
Austin, John McChord—Pittsburgh, Pennsylvania
Bradley, Orlando G.—Fort Wayne, Indiana
Buhl, Hilary J.—Fort Wayne, Indiana
Deckelman, Francis C.—Fort Wayne, Indiana
Duron, Eduardo—Fort Wayne, Indiana
Featheringham, Robert—Fort Wayne, Indiana
Gervers, William J.—Cincinnati, Ohio
Heathwood, Robert W.—College Point, New York
Herzog, Joseph F.—Pittsburgh, Pennsylvania
Kunz, Kenneth M.—St. Paul, Minnesota
Mertel, Herbert K.—Fort Wayne, Indiana
Oxman, David S.—Phoenix, Arizona
Patterson, Joe Willie—Baton Rouge, Louisiana
Ryan, Charles F.—Brooklyn, New York
Schwab, Joseph H.—New York, New York
Shepherd, David A.—Ozone Park, New York
Sherrard, Blaine O.—Charleston, Maine
Snyder, John P.—Fort Wayne, Indiana
Vilicic, Andrew F.—New York, New York
Wallace, William—New York, New York
Williams, Raymond D.—Baton Rouge, Louisiana

Change of Grade

Fellow

Miller, James A.—Santa Barbara, California

Member

Blinn, Richard P.—Tustin, California
Morison, Russel W.—Waterloo, New York
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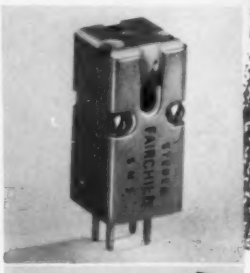

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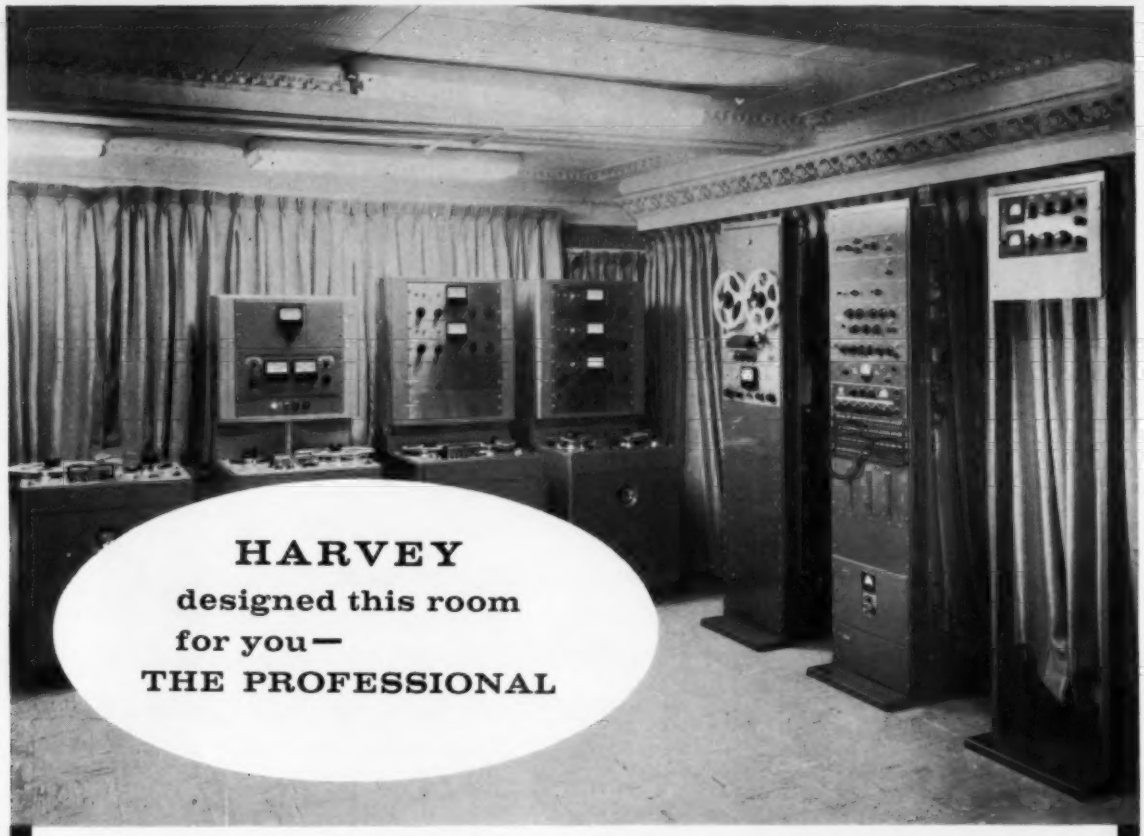

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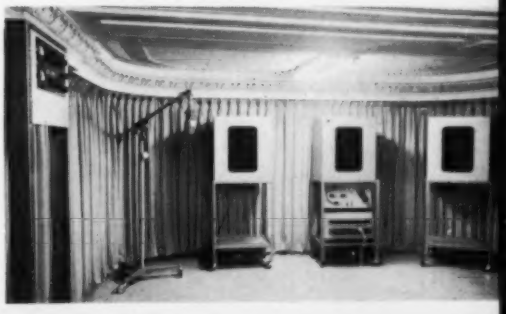

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Signal to noise ratio about 50 db

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Voltage: 0,2 millivolt for full level record

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While it is preferable for the references to be Society members, this is not indispensable.

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Information for Convention and/or JOURNAL Authors

SUGGESTIONS FOR PREPARING MANUSCRIPTS, PHOTOGRAPHS, CHARTS, DRAWINGS, AND LANTERN SLIDES; LETTERING SIZES, MAILING INSTRUCTIONS, ETC.

A Journal of original record. Papers originally published elsewhere or promised for publication elsewhere are not accepted for presentation at Audio Engineering Society Conventions or for publication in the JOURNAL OF THE AES. In rare instances, exceptions may be made by the Convention and Publications Committees at their discretion.

All papers presented at AES Conventions are automatically considered for publication in the JOURNAL.

Free and clear. It is assumed that all manuscripts submitted to the AES are offered *free and clear.* Any paper accepted for presentation before the Society or for publication in the JOURNAL OF THE AES becomes the exclusive property of the Audio Engineering Society. Complete publication rights are held by the AES for primary publication in the JOURNAL.

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Multiple bylines. If the paper is to have more than one author, the exact form of the byline should be indicated on the manuscript and will be considered correct by the AES as given.

Vital data. Make sure that the first page of your manuscript carries your name precisely as you would like it rendered in the event of publication. If your business, professional or academic affiliation is to be given, this, too, should be included. If your position bears a formal title, you may include it.

Typing the manuscript. Only one side of the sheet should be used. Margins should be at least one inch wide on each side. Triple spacing is preferred.

Reviewing of the manuscript is speeded up considerably if several copies are submitted. (This is helpful but in no sense mandatory.) The copies may be clear carbons, mimeographed sheets, copies made on a "spirit" duplicator, or blueprints. Review copies of diagrams, schematics, and graphs may be made by any convenient process.

Abstract. The author should precede his text by an abstract summarizing the paper in general. The abstract may include a summary of observations and conclusions set forth.

Subheads. Subheadings for important sections of the paper make it easier to read in printed form.

References. References to periodical literature should include the author's full name, exact title of the article or paper cited, full name of the publication, volume number, page numbers, month, and year. Book references should include the author's full name, full title of the book, the specific page or pages referred to, the publisher, place of publication, and year of publication. References to patent literature should preferably be given as follows: name, number of the patent (U. S. or foreign), date of filing, and date of issue. A brief description of the patent is helpful.

Mathematical symbols. Care should be taken to make all mathematical expressions clear to the printer. All Greek letters and any unusual symbols should be identified in the margin. Only the very simplest formulas should be typewritten: all others should be written in carefully in ink. Do not neglect to give the meanings of all symbols used.

Captions for illustrations. A caption—properly identified—should be supplied for each illustration and a legend for each chart. These captions should be listed—in complete form and consecutively—on a single sheet of paper.

Photographs. All illustrative material needed for a particular manuscript should be referred to specifically in the text and should accompany the manuscript when it is mailed.

Photographs and drawings should be prepared carefully to insure good reproduction. Photos should be standard 8 in. x 10 in. glossy prints. Since extremely fine detail tends to be obscured in all reproduction processes, it is often advisable to include a separate picture—i.e., a "closeup"—of any highly significant detail, in addition to the general view which describes the overall field.

Drawings. The *reproduction* copies of sketches, of curves, or schematics (as distinguished from any *review* copies submitted) should preferably be original drawings in India ink on white paper or on tracing paper, 8½ in. x 11 in. Curves made on conventional graph paper will reproduce poorly, but black India ink tracings, in which only the principal cross-section lines are rendered, are satisfactory.

Sharp, high-contrast photographs of line drawings are acceptable.

Please do not send black-and-white lantern slides or color transparencies for publication—only photographic prints or original drawings should be submitted.

Lettering. On the aforementioned 8½ in. x 11 in. sheets, lettering and numerical data should not be less than 0.12 in. high. Necessary labeling should be lettered onto curves and sketches. On the other hand if extensive descriptive material is needed, it is better to put such information into the typewritten captions rather than to attempt to letter the information onto the curve or sketch.

Mailing. Mail one copy of your manuscript to the Convention Chairman. This copy is for scheduling and publicity purposes. Please mail all other copies of your manuscript to the Secretary at the Society's Office.

Mail that copy of the manuscript which is accompanied by the *reproduction* copies of your illustrative material (the editor's copy) *flat*, with plenty of stiff cardboard enclosed. It is advisable to mark the envelope "PLEASE DO NOT BEND."

ORAL PRESENTATION OF THE PAPER

Time allotment. The average time allowed for any paper, unless the author requests more time on his Author's Form, is 20 minutes. If you are not going to deliver your paper in person, please supply the name of your alternate to the Convention Chairman as early as possible.

Special oral version. An informal version of approximately 20 minutes, having an air of spontaneity, is usually more effective than a rushed verbatim reading of the manuscript exactly as written for publication. Many authors, after submitting the formal full-length version of their paper for publication, prepare an *informal* version for their guidance during oral delivery.

Demonstrations. Demonstrations always add interest. They should be set up well in advance of the particular session for which they are intended and tested under actual operating conditions. The Convention Committee will cooperate in every way within its power.

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Lantern slides. Photographs, diagrams, charts, and curves intended to accompany oral delivery should be in the form of standard American lantern slides (3¼ in. x 4 in.) or 2 in. x 2 in. transparencies. The long dimension of the projected area should be horizontal in the standard slide, but may be either horizontal or vertical in the 2 in. x 2 in. slide.

PLEASE NOTE: Opaque projection of paper prints or drawings is not recommended. Nor can we guarantee that facilities for this type of projection will be provided. Because of the dimness of the image, opaque projection is unsatisfactory for large audiences.

Lettering on lantern slides. Charts, diagrams and curves from which lantern slides are to be made should be held within the proportions or overall dimensions of 7 in. x 10 in., with no more than 20, and preferably only 15 words to a chart. Vertical Gothic capitals are recommended.

For a 7 in. x 10 in. working area, the smallest desirable letter height is 0.14 in. (made with a Leroy pen 0 or Wrico pen 7).

Recommended, too, are the following line widths:

For curves—1½ to 2 points or 0.021 in. to 0.028 in.
For grid rules—½ point or 0.007 in.
For reference lines—1 point or 0.014 in.

Thumb marks. To indicate proper orientation, a thumb mark should be placed in the lower left-hand corner of the slide, when the latter is viewed as it will appear projected on the screen.

More complete information on lantern slide dimensions is given in American Standard Dimensions for Lantern Slides, Z38.7.19-1950

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